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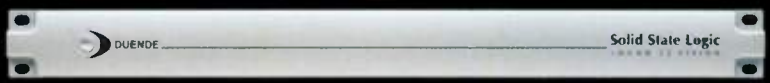

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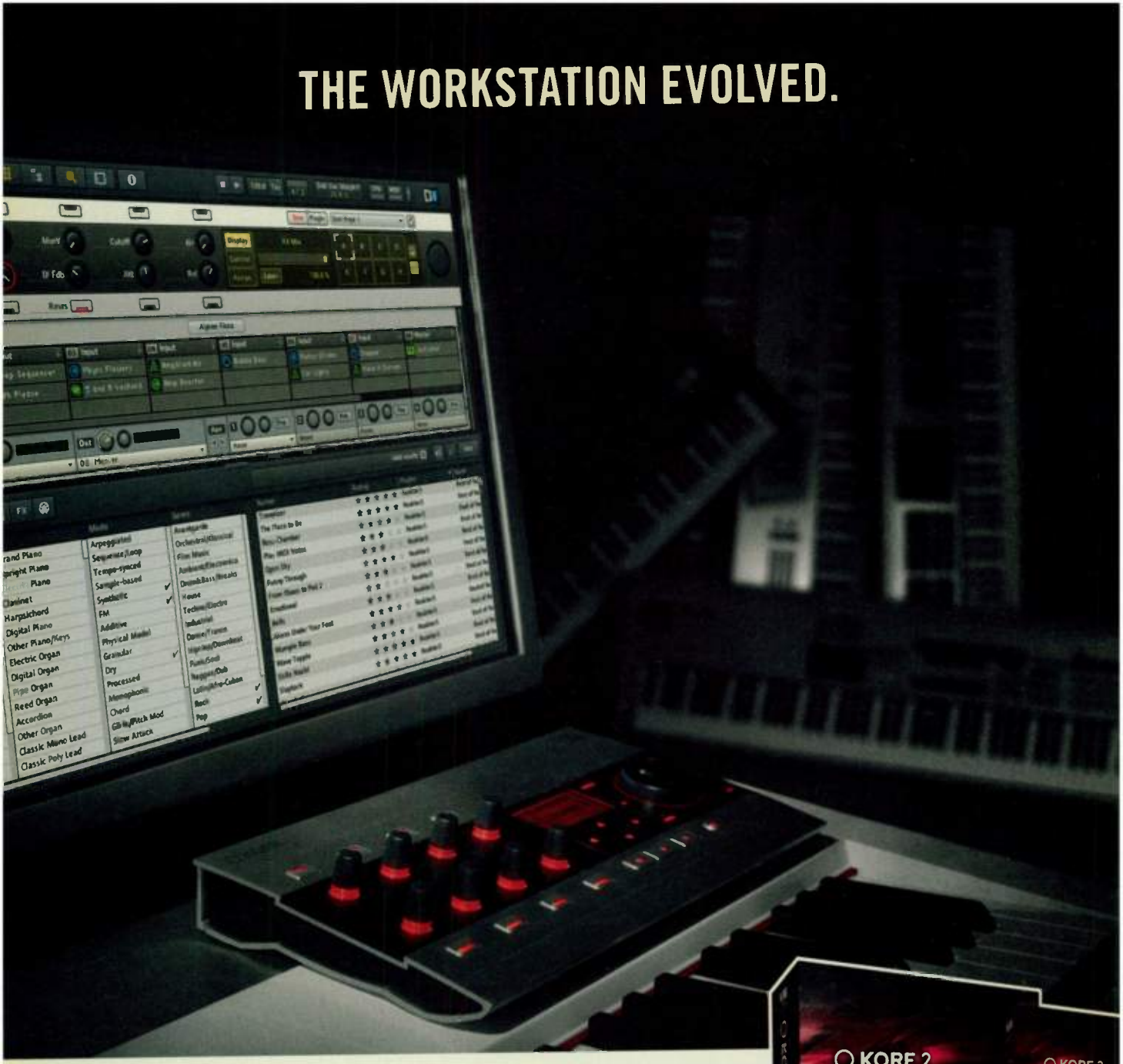

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PROFILE: M3

PROFILE: **RAY MURRAY**

Role: Producer, Organized Noize Productions.

Recent projects: Big Boi (Outkast) Solo LP, Dukes of DaVille, Dungeoneze artist Ked, Mills Gates

I'm proud of: writing and producing "Waterfalls" for TLC

I picked the M3 because: I really dig the way the M3 feels. It's got a really big, heavy sound. Not to mention the kick-ass drums!

PROFILE: **KAMAL GRAY**

Role: Keyboardist/Songwriter of The Roots

Recent Projects: new album *Rising Down*

I'm proud of: being able to put "Grammy® winning" in front of my name whenever I want... lol!

I picked the M3 because: of the touch screen for switching sounds fast. The programmable pads too because I like to play samples of myself when I'm producing tracks.

PROFILE: **LYNETTE WILLIAMS**

Role: Keyboardist

Recent projects: Keyboards for Will.i.am and on "Heat" from Mariah Carey's new release, *E=MC2 (import)*

I'm proud of: getting over the hurdle of being nervous when it comes to performing in public!

I picked the M3 because: I hook the M3 up to my computer and it's like a traveling studio. Not to mention it just looks so cool!

PROFILE: **JASON FARMER AKA J-VIBE**

Role: Keyboardist for Ky-mani Marley and Damian "Jr. Gong" Marley

Recent projects: Musical director for Ky-mani Marley on Van Halen's sold out tour

I'm proud of: Producing the hit song "Heaven Sent" for Keyshia Cole

I picked the M3 because: I can always rely on Korg to give me a real fat sound! I love being able to use the Control Faders, Motion, X-Y, and KARMA features all at the same time.

PROFILE: **ELDAR DJANGIROV**

Role: Keyboardist

Recent projects: 2007 Solo release *Reimagination*

I'm proud of: 2008 Grammy® performance & nomination

I picked the M3 because: The M3 is the best in its league, the effects are fantastic and the depth of sound exploration is limitless.

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SEPTEMBER 2008 VOL. 24, NO. 9

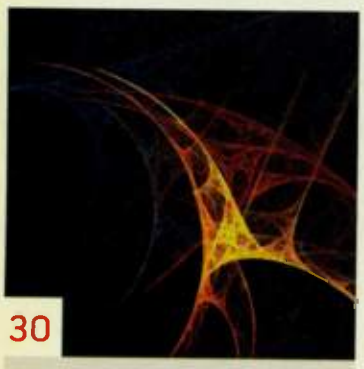
FEATURES

CHRIS DAVISON

40 DREAM PRODUCERS

The Orb's new release, *The Dream*, revisits the band's distinctive blend of dub and ambient house music. It's also the first album in years to feature both founding member Alex Paterson and bassist-producer Martin "Youth" Glover.

By Bill Murphy



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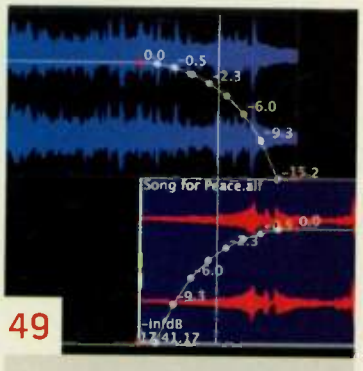
SAY IT WITH PICTURES

It's easy to turn a graphic into sound. We cover eight programs that let you produce an audio file from an image.

By Dennis Miller



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MASTER CLASS: MASTERING WITH PEAK PERFORMANCE

Explore the intricacies of mastering, including Playlist optimization, with BIAS Peak Pro 6.

By Larry the O

DEPARTMENTS

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>>QUICK PICKS 88

>> **Samson GM1U G-Track** USB microphone and interface

>> **PSPaudioware Xenon 1.0 (Mac/Win)** mastering plug-in

>> **PreSonus Monitor Station** audio controller

>>REVIEWS

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Produce – Your music, just connect the MM series to your computer via USB and use the included Cubase music production software.

For Tips & Tricks: www.motifator.com

YAMAHA

Be Redundant

"More storage equals more opportunity to lose precious memories" read a headline about disk-drive capacity at CNET's Crave Web site. The point being made was that with high-capacity drives becoming the norm in cameras and camcorders, it's easy to keep filling that drive or card to the max before off-loading the data. But by waiting until the disk is full before making a backup, you run the risk of losing everything if the drive gets corrupted, if the camera or card gets lost, or if everything falls into a lake.

Of course, musicians are in a similar situation of relying too much on onboard storage, whether they're using portable digital recorders or computers. If your entire project is on one drive and nothing is backed up, you're flirting with disaster. Although you may not have the time or resources to be sure that *all* of your data exists in three different places ("or it doesn't exist at all" goes the adage), at least back up your most important work. And be sure to annotate the backup so you'll know what's on it later.

But being prepared for problems goes beyond backing up your data. At a recent gig, one of my stompboxes died during sound check. At first it took a bit of trial and error to locate the trouble: was it a cable that failed? Or a bad wall wart, perhaps? The good news is that I had the extra cables and some batteries to test with. But once I figured out it was the echo pedal at the end of the signal chain that died, I didn't have a choice but to do without it that night.

For most people, having a completely redundant system, such as a fully stocked pedalboard waiting in the wings, is impractical. Yet how many of you keep spares of the little things in your studio or in your tour van? Do you have an extra FireWire cable? How about spare USB cables with each type of connector (the small one for your pocket recorder as well as the bigger one for your audio interface)?


JANE RICHEY

Many of you are probably rolling your eyes at this point because this is so obvious, right? But let me ask the laptop performers reading this: do you carry a spare cable to connect your controller or interface, along with a spare power supply for your computer? You have not known frustration until you've accidentally dropped and broken your power supply in some remote town on a Saturday night, far from any Best Buy or Apple store. (Your cleverly designed Max patch sits quietly on your dormant computer as you weep.)

So let's look at some obvious backup items that most of us seem to shrug off. (To save space, I will list only the things that I have personally seen people forget, both at gigs and when they visit a studio.)

- Cables. Whatever you use most, have an extra. That goes for ¼-inch, ⅛-inch, RCA/coaxial, XLR, BNC, all manner of digital cables, and even an IEC power connector.
- Strings. Guitarists must have an extra set with them, no matter where they are (stage or studio) or who they are (even famous players). Same goes for bass, banjo, mandolin, and so on.
- Batteries. Or better yet, get a modular power system for your stompboxes. But have a 9V or two around for emergencies.
- Power strip. Electronic musicians can never have too many of these. If you travel with it, put your name on it or it will walk.
- Blank media. How hard is it to keep a blank CD-R and DVD-R with you when you travel, as well as a 1 GB USB thumb drive? Always have a spool of writable media available in your studio.
- Drummers should never be too far from a drum key and an extra hi-hat clutch, snare drum head, and bass drum pedal.

You get the point.

The bottom line is that you should think ahead and prepare for the worst-case scenarios of any upcoming projects or tours. Dropping some cash on a second USB cable might seem like a dumb idea now, but you'll change your tune when it saves your late-night session or gig.

Gino Robair
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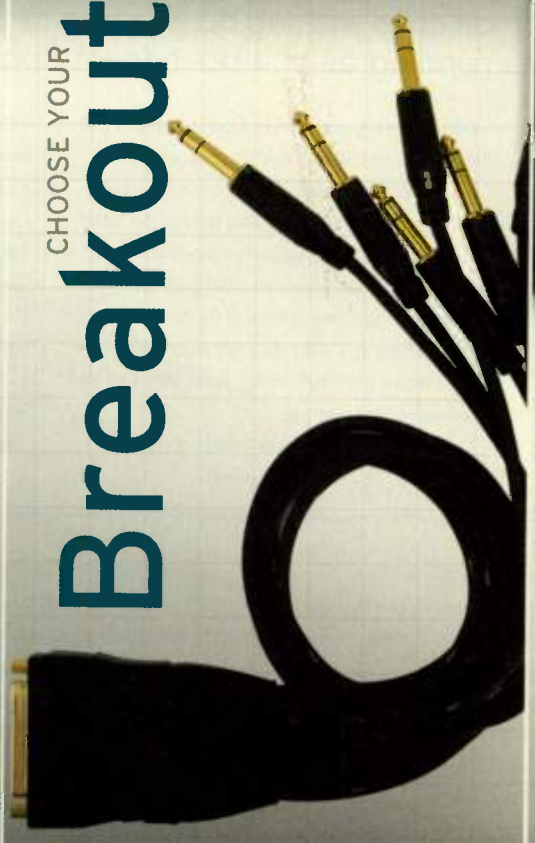

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LETTERS

Hear What's Around You

After reading Gino Robair's editorial in the July '08 issue (see "First Take"), I just wanted to thank him for bringing up a subject that is often overlooked. I own an iPod, but it more often than not stays in my project studio. My wife doesn't understand how I can sit waiting for our flight to arrive and not have some form of iPod or cell phone to entertain me. What she's missing is what I'm listening to—the delicate patter of snow on the panoramic windows. The whimper of a small child wishing to get out of her stroller. The fade-in and fade-out of the sinks running as the bathroom door opens and closes.

This kind of situational awareness aids in all aspects of recording and production. Listening to an environment enables an engineer to understand stereo and 5.1 panning. It allows a sound designer to more intimately acquaint himself with what the audience will react to in a movie. It frees one from being trapped in recorded music, so the ears are refreshed.

There is another aspect to listening that I'd like to touch on: listening to what you've already been listening to. If you're a live-sound engineer, go to a concert and just listen and enjoy. If you're a recording engineer, go to a tracking session and just watch and listen. I'm a drummer as well as a studio and live-sound producer, so I have the unique advantage of being able to hear the "back" of a mix: the way the speakers echo off the walls, the way the drum kit sounds sitting behind it rather than standing in front of it. These sound observations and many more are essential to the way I record and create music. Every time I sit back and listen, I'm learning.

Before you spend hours in front of the TV watching instructional DVDs in an effort to create amazing soundscapes, try listening to an amazing soundscape. Like an ocean cresting at full tide. Listen to the seagulls above and to the left of you. The center-channel presence of the crashing waves. The distortion of the relentless breeze in your ears. The flapping of a kite behind and to the left of your position. There are sounds all around.

Every second, we are listening in 360 [degrees]. How can this translate to your stereo or 5.1 mix? How could this affect your miking positions?

—JOSH CONTI, VIA EMAIL

MASTERING A TOUCHY SUBJECT

Thank you for your article "Production Values: Keeping It Dynamic" (see the July 2008 issue) on mastering engineer Bob Katz. I cannot tell you the number of projects I have encountered that have been robbed of their character during mastering. Music should be alive and dynamic. We should hear it like the giggle of a baby and the roar of a lion. They are not the same and they should not be squashed together to sound the same, especially for the purpose of making the average volume louder. If the listener wants it louder, "They make a knob for that."

The clients I work with—newcomers and indie artists—are especially vulnerable to this problem. The process is typically new to them, and they don't understand the purpose of mastering to begin with. They rarely have the ear or the tenacity to reject overcompression. Their projects suffer as a result.

So I say to the new and the experienced, the garage band and the studio exec: insist that your mastering studio balance your music without asphyxiating it.

RICHARD F. SAMS
VIA EMAIL

Mike Levine's interview with mastering engineer Bob Katz was an eye-opener. Regarding the loudness wars on CD, I was in Bob's camp for the longest time until the interview forced me to reconsider the reality of most listening spaces. The loudness wars are actually born from the pervasive use of compact technology, where it's now possible to listen in hi-fi literally anywhere. I was buying his argument all the way until his dynamics chart (see Fig. 4 on p. 42) listed the cinema as a typical listening space. How about at work, or at poolside with kids screaming?

The CD loudness wars weren't created by peak-normalize ability, but by stuff like my MP3 player and the resulting need to have the music cut through the clutter. Sure, it sucks if you're mixing pop music all day long in a perfect room where the dynamic range would be wasted. I feel for Bob and his cadre of enlightened clients, but his reality is way different from mine.

JOHNNY HUNKINS
VIA EMAIL

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MICROPHONES

FRONTPANEL By Gino Robair

Download of the Month

Illformed dblue Glitch 1.3.04 (Win) By Len Sasso

Glitch is a free VST effects plug-in for Windows created by Kieran Foster (aka dblue). You create a step pattern up to four bars long with step sizes ranging from a 16th note to the full pattern length. You then select one of nine effects for each step and let Glitch do its work. Instead of selecting an effect, you can leave a step empty or let the plug-in choose an effect at random. Use the mouse, MIDI remote control, or host automation to select patterns on the fly from Glitch's 16-pattern memory (see **Web Clip 1**).

Setting up patterns and programming the effects is so straightforward that you don't need a manual—the help comments that appear when you mouse over the controls are sufficient. The effects are TapeStop (tape-style speedup and slowdown), Modulator (FM), Retrigger, Shuffler, Reverser, Crusher, Gater, Delay, and Stretcher (granular time-stretching). Each effect has its own multimode filter and mixer controls. Beyond that, you get global controls for fading each step, overdrive, and a master multimode filter.

ONLINE BONUS MATERIAL

Glitch is the type of processor that begs for randomization, and you have plenty of options. A single click will randomize the step pattern, the effect selected for each step, or both. Each effect has its own random button as well as a probability numerical to set how likely the effect is to be selected for a random step. For example, to create a fill in the last two beats of a 2-bar pattern using a Shuffler or Retrigger module, insert a 2-beat random effect, set the Shuffler and Retrigger modules' probabilities to 100, and set all other modules' probabilities to 0 (see **Web Clip 2**).

Anything from drums to comping to vocals is good source material for Glitch. It's great for processing audio on the fly, and you can make it as subtle or aggressive as you choose. The Illformed Web site (illformed.org) features audio examples, a video demo, and a users blog, as well as the Glitch download.

OPTION-CLICK By David Battino

Harlan Hogan's collapsible Porta-Booth provides tight vocals to go.

The Closet Vocalist

Discover cool sounds lurking inside the wardrobe

Whenever I get compliments on the sound of my Podcast (digitalmedia.oreilly.com/podcast-audio), I like to blow the listener's mind by revealing *how* I record the narration. No pricey studio, exotic soundproofing, or even pop filters—I simply hang a mic between the shirts in my closet, speak across the diaphragm instead of directly into it, and then compress and EQ the recording with iZotope Ozone. The shirts reduce slapback, the ugly fingerprint of home studios. Speaking at an angle minimizes *p* pops. And Ozone beefs up the level of the USB mics I typically use: a Røde Podcaster and an sE Electronics USB2200a.

I hit on the closet approach while trying to escape my computer's howling fans, but later learned that voice-over pro Harlan Hogan used to build forts out of hotel couch cushions to record his on-the-go vocals. See harlanhogan.com for his latest brainstorm, the ingenious Porta-Booth. (For more about David Battino's work, visit batmosphere.com.)

Georg Neumann: Milestones

1923

Reisz Microphone
Georg Neumann's carbon design greatly improved on the standard mics of the day.

1928
CMV3
The first commercially manufactured condenser mic, the Neumann Bottle. Later, the CMV3a had a pair of interchangeable capsules.

1947

Sealed NiCad Battery
Neumann's gas-tight, nickel-cadmium accumulator battery led to the creation of the 1.5V Stabilyte cell.

1947
U47
This condenser mic featured the first switchable polar patterns (cardioid and omni) thanks to its dual-diaphragm design.

THIS MONTH'S SOUNDTRACK

These albums encompass a diverse range of styles and composition methods, from pop and electronica to space music and experimental.

1. Portishead: *Third* (Mercury)
2. Furt plus: *Equals* (PSI)
3. Robert Rich/Ian Boddy: *React* (DIN)
4. Matmos: *Supreme Balloon* (Matador)
5. The League of Automatic Music Composers: *1978–1983* (New World)


EVA VERMANDEL

1.

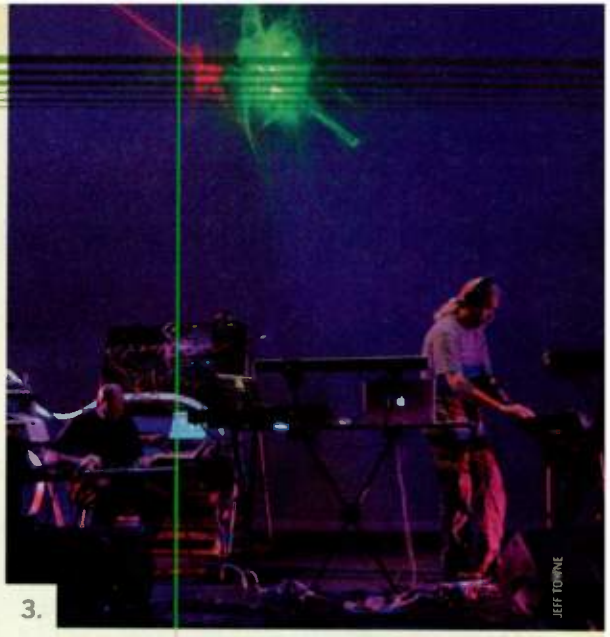
PORTISHEAD
Strange, often dreamy, but always pushing the boundaries of pop music in the right directions.

FURT PLUS
Acoustic instrumentalists are subjected to extreme sampling and processing by this impressive English duo.

ROBERT RICH/IAN BODDY
A concert recording that demonstrates why these two are masters of electronic music and sound design.

MATMOS
A lighthearted disc composed using Cycling '74 Max and vintage hardware synths.

THE LEAGUE OF AUTOMATIC MUSIC COMPOSERS
Computer network pioneers with a chiptune and glitch aesthetic.


JEFF TOWNE

3.


AJ FARKAS

4.


JIM NEWBERRY

2.


PETER ABRAMOWITSCH

5.

EM POLL
What sampler software do you use the most? a) Ableton Sampler, b) Apple EXS24, c) Digidesign Structure, d) E-mu Emulator, e) MOTU Mach Five, f) Native Instruments Kontakt, g) Propellerhead NN-XT, h) Steinberg HALion, i) Tascam GigaStudio, j) Yellow Tools Independence, k) other.
Submit your answer to this poll and others at emusician.com. This is not a scientific poll but a tabulation of readers' responses and is just for fun!

1956
Stereo Recording Lathe
Developed jointly with Teldec, it combined the AM32a lathe with a newly designed ZS90/45 cutting head.

1966
48V Phantom Power
Although other schemes existed before, Neumann created the standard.

1966
KM84 Microphone
This was the first commercial mic to use 48V phantom power.

By Len Sasso

AKAI XR20

BEATS TO GO

The new XR20 Beat Production Center ($499 [MSRP]) from Akai (akaipro.com) is a desktop unit with a sound set designed especially for hip-hop and R&B. Its more than 700 sounds by Chronic Music range from one-shot drum hits to bass and synth sounds. You get 99 presets and 99 user patterns, backlit pads that follow the beat, a mic input for mixing vocals over patterns, and a headphone output. In Pattern Play mode, pads trigger patterns rather than sounds. A large, backlit LCD lets you know what's going on. You can power the XR20 with batteries or an AC adapter.

SOUND DEVICES 788T RECORDER

CAUGHT IN THE ACT

With the release of the 788T digital recorder ($5,995 [MSRP]), Sound Devices (sounddevices.com) expands its 7-series line of digital audio recorders to eight tracks and adds a host of new features. The 10×1.65×6.25–inch unit weighs less than 4 pounds and sports a 160 GB hard drive, a CompactFlash slot, and FireWire storage. FireWire and USB connections are provided for data transfer. The 788T has eight phantom-powered, LED-metered mic inputs, along with digital I/O and a headphone output. In addition to its full-featured timecode clock, the unit offers word-clock I/O and trilevel video sync input. You can use external power or removable, rechargeable Li-ion cells.

TC ELECTRONIC DESKTOP KONNEKT 6

TC Electronic (tcelectronic.com) has just released the latest in its Konnekt series of audio interfaces. The compact Desktop Konnekt 6 (Mac/Win, $249) offers the same quality and features as the rest of the line, including the Big Control volume knob, Impact mic preamps with 48V phantom power, high-resolution LED metering, and separate headphone level and source. You get mic/instrument, instrument/instrument (true high-impedance guitar), and stereo line inputs, along with balanced stereo outputs. The built-in M40 Studio Reverb is also supplied as a VST and AU plug-in. The unit is FireWire bus powered and supports 24-bit, 192 kHz audio.

GET KONNEKTED

Cycling '74 Max 5

The latest incarnation of Max/MSP (Mac/Win, $495; with Jitter, $699 [both MSRP]) from Cycling '74 (cycling74.com) boasts a variety of new features aimed at giving increased flexibility and greater ease of use. Most notably, the Patcher has been enhanced with multiple undos, a pop-up object palette, and simultaneous open views. Presentation mode lets you arrange objects in the user interface independently of their size and arrangement in the Patcher. Integrated documentation, message help, and debugging tools make Max patching almost intuitive, and there are numerous examples and tutorials to get you started.

MAXED OUT

FAW CIRCLE

Circle (Mac/Win, $199) is the first product from Irish software developer Future Audio Workshop (futureaudioworkshop.com). This standalone and plug-in (VST, AU, and RTAS) soft synth stretches the analog-modeled paradigm in various ways while placing primary emphasis on ease of use. Its four oscillators offer waveform morphing between 110 single-cycle waveforms. Those, together with noise and a tuned-delay feedback circuit, are mixed into a 3-bank filter and distortion processor. Drag-and-drop modulation routing from five multimode modulators (each of which is configurable as an LFO, an ADSR envelope, or a step sequencer) makes complex setups a snap. And you get 500 factory presets to show you the way.

JOIN THE CIRCLE

Sound Advice

Ableton's Orchestral Instruments

Orchestral Instrument Collection (Mac/Win, $599) is a bundle for Ableton Live 7 (ableton.com) of four orchestral instrument families: strings, brass, woodwinds, and percussion. The collections are also available individually for $189. Produced in partnership with Sonivox, this 25 GB collection of multisampled Simpler instruments embedded in Live's Instrument Racks with premapped Macro controls offers a variety of articulations, which are available either as individual presets or combined for real-time switching. All instruments come in 16-bit LE and 24-bit full and lite versions. CPU and memory resources are managed under the hood with Live's SmartPriming technology to maximize performance.

Big Fish Audio's Found Percussion

Found Percussion ($99.95) is the second Big Fish Audio (bigfishaudio.com) release to feature percussion construction kits (85) made of sounds produced from everyday objects. Like the earlier *Tool Shed Percussion*, this 6 GB collection includes 4- and 8-bar loops in Acidized WAV, REX, and Apple Loops formats that are organized in kits ranging in tempo from 49 to 190 bpm. The tempo-flexible formats let you easily mix and match loops from different kits, and each kit has one or more composite loops to use as a bed. Titles like Mallets on a Storage Bin and Slapped Coffee Tin tell you where this collection is headed.

PowerFX's All Them Beats

In the liner notes to *All Them Beats* ($69), PowerFX (powerfx.com) founder Bil Bryant recounts his first gig, at 16, as a dance-band drummer required to instantly recall countless rhythms. Fast-forward to this cross-genre collection of drum loops in 18 categories ranging from world (Africa, Caribbean, Europe, IndoAsian, Latin) to classic (Blues, Country, Jazz, New Orleans) to modern (Electronica, Hip Hop, Rock, FunkSoul). The intent is to use these with your own kits, so the main vehicle is Standard MIDI Files, but each loop is also provided in WAV and REX audio formats.

HUMANOID SOUND SYSTEMS SCANNED SYNTH PRO 2

Scanned Synth Pro 2 (Win, $99.99) is the first commercial implementation of a variation on wavetable synthesis, heretofore found only on synths created in do-it-yourself environments such as Csound. It produces sounds reminiscent of both wavetable synthesis and physical modeling: edgy, ethereal, and unpredictable. This VSTi plug-in adds standard multimode filters and effects, along with ADSR envelopes and LFOs for modulation. A customizable randomizing scheme nearly always generates usable variations on the many factory presets. Buy it or download the free demo and scaled-down versions from Humanoid Sound Systems (humanoidsounds.co.uk).

WAVE MECHANICS

AUDIO DAMAGE KOMBINAT

TRIPLE THREAT

Audio Damage (audiodamage.com) proudly proclaims, "We're In Yr DAW Murderin' Yr Bitz," and the release of Kombinat (Mac/Win, $49) won't tarnish that image. It's a 3-band audio mangler with seven distortion engines: fuzz, saturation, distortion, clipping, warping, bit reduction, and ring modulation. The bands, which you can arrange in series or parallel, are preceded by a 3-band EQ with adjustable crossover, and in parallel mode, each band gets its own distortion engine. The signal path ends in a resonant lowpass filter and a compressor. VST versions are provided on both platforms, and AU is provided for the Mac. em

COURTESY TIM FITE

TIM FITE

Home base: Brooklyn

Primary software: Digidesign Pro Tools LE

Field-recording mic: Earthworks M30 (pair)

Web sites: timfite.com, myspace.com/timfite

Fair Ain't Fair

Fite's Flight of Fancy

Fireworks, fast-forwarding tape, and fake mandolin are all heard on Tim Fite's latest.

There's a difference between a field-recorded sound that's "to die for" and one that almost gets you killed while gathering it. Just ask Tim Fite, who, along with his friend Rob Badenoch, had a close call while sampling the sound of fireworks for use on Fite's new CD, *Fair Ain't Fair* (Anti, 2008). "The fireworks," says Fite, "malfunctioned and came shooting right at us." Luckily, no damage was done and they got a good-sounding recording of the incident through a Crown Sass-P condenser mic into a flash-card-based stereo recorder. "Even though we almost got killed, at least it was recorded in stereo," Fite says.

By Diane Gershuny

The rest of the sampling for the new CD was not nearly as hazardous, which is fortunate considering the integral role played by sampled audio throughout the production. As on previous releases *Gone Ain't Gone* (Anti, 2005) and *Over the Counter Culture* (a Web-only offering, 2007), many of the songs on *Fair Ain't Fair* are based around a sample—either self-created or pilfered from dollar-bin CDs or other friends' music. Working from his Brooklyn studio, which comprises an Apple MacBook Pro laptop loaded with Digidesign Pro Tools LE software, a Digidesign Mbox 2 interface, a Mackie Onyx mixer, a Roland Cube amp, and MXL 2001 and Groove Tubes GT67 mics, Fite built his tracks with the samples and a palette of quirky instrumentation.

Another field-recording session for the CD found Fite back at his old high school's choral rehearsal room with Badenoch and drummer Justin Riddle, where they recorded drums after hours into a PC laptop running Sony Creative Software's Vegas Pro. Badenoch put up a pair of Earthworks M30 mics mounted high in an XY pattern on a Schneider Disc stereo microphone mount to capture the room sound. Those tracks became the foundation for many of the songs.

"On 'Rats and Rags,' I was thinking about how the basis of hip-hop songs is often the drums," says Fite. "I wanted to strip this notion down, letting them [the drums] develop over the course of the track. Once I had the drums, I plugged in an old MIDI controller and began laying in rudimentary licks of fake cello and doubled that with electric guitar. I enlisted violist Marla Hansen [of My Brightest Diamond] to double the MIDI cello, and we built out the finale with overdub arrangements and vocals using the GT67 tube mic in both omni and cardioid patterns."

"Names of All the Animals" was similarly built, with the addition of a 15-minute spontaneous "noise session." "We walked in a circle counterclockwise around the mic using whatever to make noise—percussion, drums, pianos, whistles, upright bass, a megaphone, duct tape, water, wood, metal, skin. If you listen closely," says Fite, "it makes for some spectacular headphone time. In addition, we added some fake mandolin [made with a capoed guitar and heavy delay], acoustic guitar, bells, and electronic enhancement."

On "Trouble," Fite enlisted his brother, Dr. Leisure, wielding a handheld tape recorder like a DJ would use turntables. "He fast-forwarded and reversed organic sounds to create organized cacophony. There's a rip to his tones that I've never heard anywhere else."

Fite confesses that this record is more analytical than his others. "[I've always] had a firm concept before I started. For this one, I overscrutinized every sound. I had to buy the shoes too big and let the toes grow into them." em

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info@antarestech.com | www.antarestech.com

RICK ENGLISH

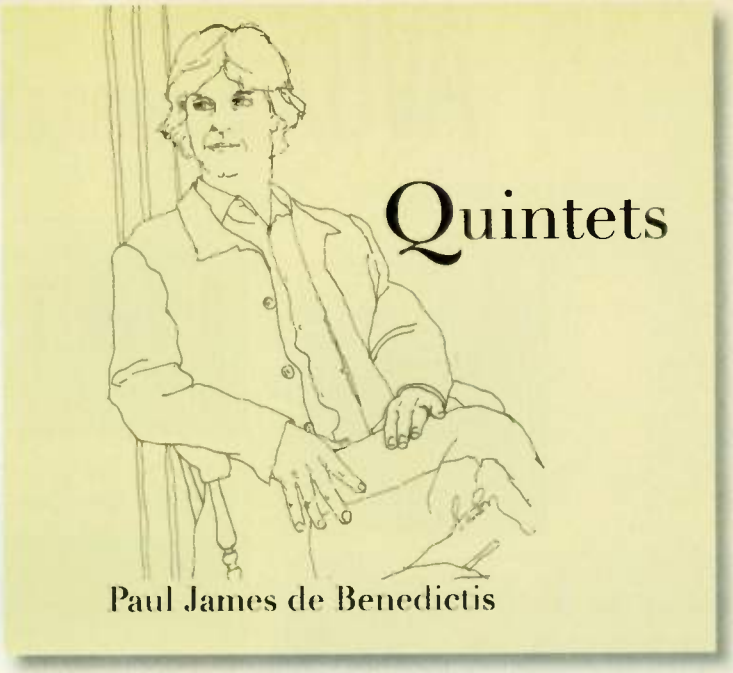
PAUL JAMES DE BENEDICTIS

Home base: Menlo Park, California

Software used: Digidesign Pro Tools, Apple Logic Pro, Opcode Studio Vision and Overture

Key hardware: Digidesign 192 I/O, Dangerous Music 2-Bus

Web sites: www.pdbmusic.com, www.mnemonicrecords.net



»» *Quintets*

All in Good Time

Paul James de Benedictis produces *Quintets* over a ten-year span

It's not often that you hear the likes of Branford Marsalis or Jeff Lorber jamming to the accompaniment of Russian string players—in surround sound, no less—but composer Paul James de Benedictis's sublime new album, *Quintets* (Mnemonic, 2008), offers just such a musical fusion.

By Emile Menasché

The idea behind *Quintets*—which de Benedictis recorded over a period of ten years using a variety of analog and digital gear—was to write chamber-music parts for string quartet and create a looser framework for soloists to interpret and improvise their parts. "I chose the musicians first," he says. "I started writing parts for each person after I got their yes."

At first, de Benedictis intended to play the piano parts himself. But "my improvised solos were not what I was hearing for the music," he says. And rather than try to cobble together a performance with technology, he thought of who could play the parts and found friends who said they would do it.

Fortunately, de Benedictis accumulated some talented friends during his years at Opcode Systems, the California-based company that created Studio Vision, the first digital audio sequencer. De Benedictis was Opcode's first employee, and over the years, his day job brought "Paul dB" into contact with many top musicians, including Marsalis, Lorber, Michael Lang, and others.

When the *Quintets* project was getting under way in the late 1990s, de Benedictis was using Studio Vision (with Digidesign Pro Tools TDM hardware) and an E-mu Proteus 2 synth module to compose, creating printable scores in Opcode's notation program, Overture. "My current rig consists of a dual 2 GHz Mac G5 and a small Pro Tools HD setup with two DSP Farms, a 192 I/O, and a Dangerous Music 2-Bus analog [summing amplifier] for summing and for mixing back into Pro Tools," says de Benedictis, who points out that Opcode's Dave Oppenheim helped enhance Pro Tools' MIDI features. "I also use Apple Logic for some composing, especially for the notation view."

When de Benedictis brought the printed scores to the Arlekin String Quartet, he discovered the limitations of composing with sampled sounds. "By sketching the string quartets out and using the synthesizers as a crutch, I forced violins to interact in ways that don't work as well when they play live and have to listen to each other," he explains. Enlisting another friend, former Frank Zappa musical director Scott Thunes, de Benedictis altered the score to be more playable.

Once the arrangements were working, de Benedictis brought in Tom Carr at the Music Annex in Menlo Park, California, to record the strings. Carr used six mics—one for each string player and two room mics—through a Neve console, tracking to 24-track analog tape with Dolby SR for maximum sound quality (one track was striped with SMPTE for syncing to DAWs). The solo instruments were recorded at various facilities, including Lorber's, Marsalis's, and de Benedictis's respective home studios, and—at the behest of pianist Lang—at a Fazioli piano dealership with an on-site studio.

The remote tracks were eventually printed to the 24-track tape before being transferred to Steinberg Nuendo at 24-bit, 96 kHz for the stereo and surround mixes, which were engineered by Michael Romanowski. "The mixes in the surround are what I was hearing when I conceived the piece," de Benedictis says. "There's a lot going on—it's intricate—and the surround really makes a difference."

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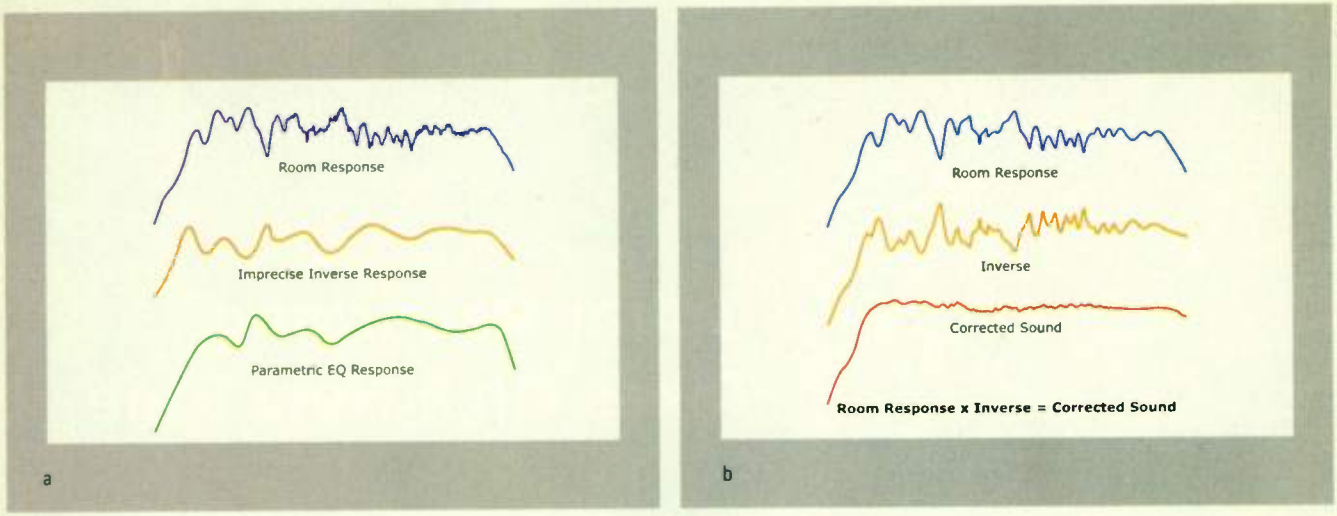


COURTESY AUDYSSEY LABORATORIES

FIG. 1: A parametric EQ can go only so far in correcting room anomalies (a). MultEQ creates a far more precise filter that is the inverse of the room's measured response, resulting in a much flatter final response (b).

Got Modes?

A new technology to combat room-based distortion. | By Scott Wilkinson

As all electronic musicians know, reproducing sound from speakers in most rooms can be fraught with problems. In particular, the sound reflects from the walls, ceiling, and floor, interacting with the sound coming directly from the speakers as well as other reflections. These interactions can seriously alter the spectral profile of any sound emanating from the speakers, causing what you hear to be distorted.

In addition, the distortion is different depending on where in the room you are located. For example, standing waves occur at frequencies whose wavelengths are simple multiples of a room's dimensions, causing nulls at some locations and peaks at others. These so-called room modes are the bane of all recording engineers and the main reason that recording studios are so carefully designed.

For many years, acousticians have used graphic and parametric equalizers to correct such problems, which can persist even after the room has been acoustically treated. But these remedies go only so far. In most cases, conventional equalizers don't have enough bands, and the width of each band in a graphic EQ is not controllable. As a result, the correction is less than perfect (see Fig. 1a).

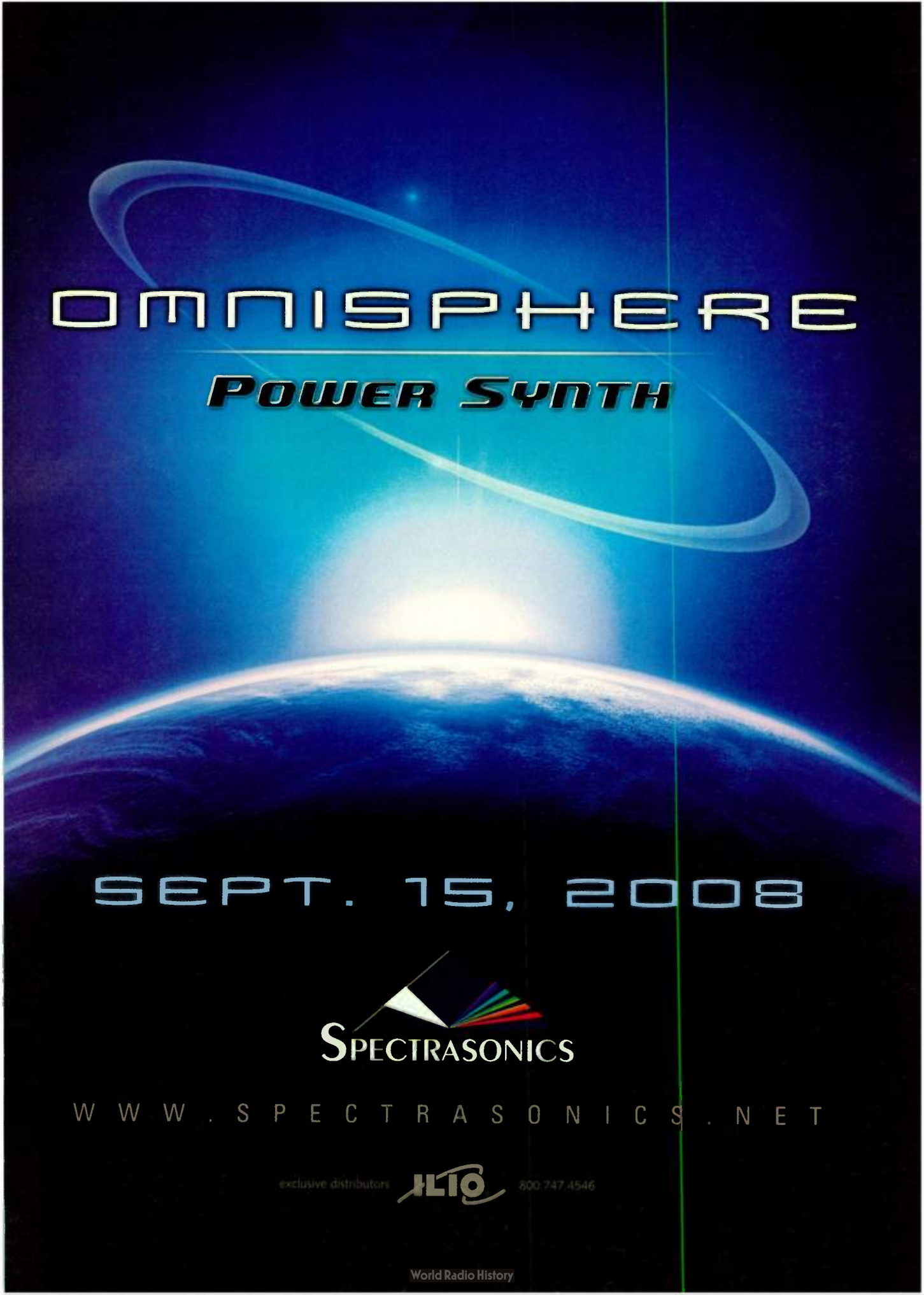
A company named Audyssey Laboratories (audyssey.com) has come up with a different approach to room correction that goes much further than conventional solutions. The system, dubbed MultEQ, began as an academic research project at the University of Southern California under the direction of professor Chris Kyriakakis and one of his PhD students, Sunil Bharitkar. In 2002 they went on to found Audyssey with USC Audio Lab researcher Phil Hilmes and USC professor Tomlinson Holman (of THX fame).

The basic idea is to play test tones through the sound system in a room and measure the room's acoustic response at multiple listening positions. MultEQ then creates a precisely tuned inverse filter that counteracts most room-based distortions (see Fig. 1b). The technology, rather than using a few EQ bands, uses finite impulse response (FIR) filters with hundreds of control points, allowing a much more accurate and comprehensive correction in the time and frequency domains. A majority of the system's correction power is dedicated to the low end of the frequency spectrum, because that's where the worst acoustic anomalies arise.

Instead of simple averaging, the MultEQ algorithm accounts for the differences at each listening position using a technique called *fuzzy clustering*, which assigns locations with similar responses to a group or cluster. The process also calculates weighting factors that reflect the relative importance of each location to the other clusters. This allows the system to devise one filter that corrects the anomalies at various locations over a large listening area. The number of measurable locations is limited only by the amount of memory and DSP horsepower available.

MultEQ has been implemented in many modern A/V receivers to automatically calibrate the sound for up to eight locations in a consumer's listening room, and it has enjoyed accolades from many reviewers. Now Audyssey hopes to make similar inroads with recording studios. Two products are currently available: a hardware processor from Audyssey called the Sound Equalizer (which requires a Windows PC to provide the user interface) and a software plug-in from IK Multimedia called Advanced Room Correction (ARC).

I've heard the result of MultEQ in consumer systems, and it's very impressive. The sound is clean and clear, and the difference between turning it off and on is like night and day. This system could be a real boon for recording musicians who want to hear their music exactly as they intend it, without the unwanted contributions from their rooms. em



SAY IT WITH PICTURES

Eight programs that convert images to music.

By Dennis Miller

Everybody knows that a picture is worth a thousand words. But did you know that a picture can also buy you the frequencies of a thousand oscillators? Using any of several modern music applications, you can convert the data that forms a picture on the computer screen into information that will generate an audio file. In most cases, the program uses the vertical position of each pixel of the image to control frequency and one or more color (RGB) values for parameters such as amplitude and stereo position.

In this article, I'll look at a number of programs for both Mac and Windows that allow you to perform this alchemy. Included are Adobe Audition 3, Thomas Baudel's HighC 2.2, Camel Audio's Cameleon 5000 1.5, Rasmus Ekman's CoagulaLight 1.66, Nicolas Fournel's AudioPaint 2.1, Image Line FL Studio 8, U&I Software MetaSynth 4, and VirSyn Poseidon 1.4. I'll also cover Mark Coniglio's Isadora 1.2.9, which can convert images to sound but is better suited to work in the opposite direction (see the sidebar "Izzy Gets Down"). I'll start with a general overview of the field before looking at each program individually.

Note that AudioPaint, Coagula, and HighC are standalone applications whose only role in life is the conversion of images to sound and, perhaps, vice versa. For Audition, Cameleon, FL Studio, MetaSynth, and Poseidon, image-to-sound conversion is just one of many features. Also, many of the ideas that drive image-to-sound software stem from research by Iannis Xenakis, a Greek composer whose UPIC system was among the first to allow musicians to draw the data used to generate sounds (see the **online bonus material** "The UPIC System" at emusician.com). Xenakis used imagery in creative ways when composing both his acoustic and electronic works.

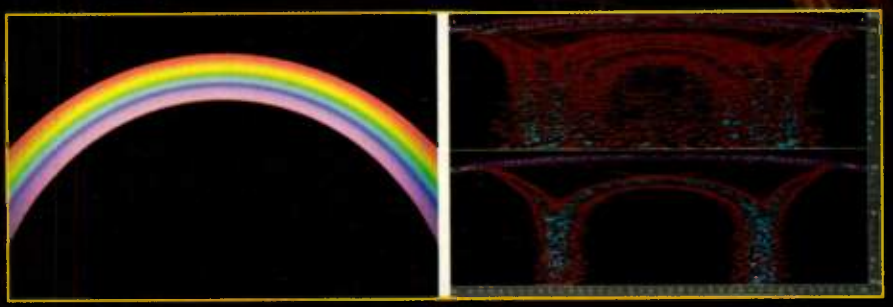


FIG. 1: This image shows the original bitmap (left) that Audition used to produce the spectrum on the right.

ONLINE BONUS MATERIAL

Roundabout

All of the programs in this roundup share the ability to convert an image into an audio file, but the range of editing and processing features you'll find varies widely. For example, all except HighC allow you to import a preexisting bitmap image (typically a BMP or PICT file, but in some cases other formats as well), and all but Cameleon, FL Studio, and Poseidon let you modify or process an image before using it to generate a sound file. Audition, FL Studio, MetaSynth, and Poseidon convert the graphics file into a 2-D sonogram display, which shows frequency on the *y*-axis and time on the *x*-axis and uses intensity (brightness) to represent amplitude (see Fig. 1). HighC uses a hybrid musical-score/piano-roll metaphor and, like the original UPIC system, provides tools for drawing the gestures and shapes that will control musical parameters. Coagula and MetaSynth also provide tools for drawing an image from scratch, while AudioPaint lets you generate a new image automatically with its configurable Lines & Curves and Clouds of Points tools.

Deciding how to extract data from the bitmap and how to use the extracted data is a big part of these programs' toolkits. Poseidon takes each pixel on the *y*-axis of the image and assigns it a frequency value within the range of 20 Hz to 22.05 kHz, then uses additive synthesis to

generate a new sound from the sum of those values. The amplitude of each partial changes over time depending on the brightness of the pixel, and each pixel accounts for about 3 ms of the new sound's duration. Most of the other programs work in a similar manner, though several—for example, AudioPaint and Audition—let you choose how the program will interpolate from one amplitude value to the next. FL Studio lets you specify an arbitrary number of partials (up to 999) for the resynthesis regardless of the size of the original image, and AudioPaint, Audition, Coagula, FL Studio, and MetaSynth let you set an arbitrary frequency range over which new partials will be generated.

Some of the programs—for instance, AudioPaint, Audition, and FL Studio—let you determine whether the new partials are distributed in a linear or log fashion (linear distributes the frequencies in increments of hertz, while log uses increments of cents). MetaSynth goes quite a bit further by including a large number of tuning options. You could, for example, space the partials of the new sound in steps of different types of traditional scales (whole tone, major/minor, and so on), using various microtonal increments (from 4 to 1,024 divisions to the octave, with more than 1,000 scales included), or, like AudioPaint and Poseidon, using scales in Scala format. (Currently over 4,000 different scales are available at the Scala Web site, xs4all.nl/~huygensf/scala/.) HighC lets you create your own pitch/frequency scales, but you have to create a list of tuning increments by typing it. (I'll discuss the other programs, including the additional extraction parameters they offer, in their respective sections.)

In addition to using additive to generate the new sound, FL Studio offers a nonadjustable form of granular synthesis. HighC and MetaSynth can use additive along with several other synthesis methods, including granular and FM. AudioPaint and MetaSynth will create a new sound by using the extracted data to control the playback parameters of samples (pitch-shifting and time-stretching, for instance).

Cameleon can display the converted image as an editable two-dimensional (frequency and amplitude) spectral plot and offers handy tools to manipulate the newly generated harmonic spectrum prior to synthesizing the sound. It also has the ability to use information extracted from an image to generate discrete bands of noise and allows you to alter the amplitudes of both the harmonic partials and noise components individually (see Fig. 2). Audition also lets you add a bit of random offset to the individual frequency components as part of the conversion process.

It's Not How Long You Make It

You can set the duration of the new file to any arbitrary length (with limits in some cases) in AudioPaint, Coagula, FL Studio, HighC, and MetaSynth, then once the program works its magic, all except AudioPaint, Coagula, and HighC allow you to manipulate the new audio file in various ways. As a full-featured sound-design "workstation," MetaSynth offers an especially large number of effects and processes for that purpose, while Cameleon and Poseidon provide the professional processing and performance tools you'd expect from high-quality modern soft synths. Audition also has a wide range of options for working with your new audio file and is the only program in this group to provide a traditional multitrack-audio interface.

All of the programs allow you to export your new audio file, with 16-bit WAV being the most commonly supported format. And of the programs that can show a sonogram display of the converted image file, all but Poseidon let you export the display as a graphic image, perhaps for external processing and reimporting (FL Studio lets you copy the image and paste it into an external image editor).

FIG. 2: Cameleon displays the converted image using adjustable sliders that represent the amplitude for each partial (top). A separate set of sliders is available for adjusting the noise components (bottom).

Documentation varies widely among these programs, with Audition (which supplies the only printed documentation) and MetaSynth offering the best in the class. HighC provides some useful getting-started tutorials; AudioPaint, a short getting-started PDF; and Coagula, a thorough online help system (you can open the HLP file directly on the desktop). You'll also find users forums for a number of the programs, some of which are more active than others.

Keep in mind that in most cases, you won't automatically get musically useful results from any random image you choose to convert, regardless of which program you're using. I found that a bit of parameter tweaking, a fair amount of "postprocessing" (reverb, pitch-shifting, and the like), and, above all, a lot of trial and error were often needed for me to get something I could use. Also, the programs that allow you to draw gestures that will control musical parameters tend to be far more useful and, ultimately, satisfying. That's probably no surprise given the long tradition of using graphic symbols to specify musical parameters in Western music.

Adobe Audition 3 (Win, $349)

Audition is a full-featured stereo and multitrack audio editor, and given its family tree, which includes video and graphics apps Photoshop, Premiere Pro, and After Effects, it's no surprise that it boasts some innovative approaches to working with images. Audition's Spectral Frequency Display is where you work with spectra you generate both by analyzing an audio file and by importing graphic images, and the features it offers in both cases are identical.

By default, the program maps an imported image using increments of either 100 Hz per pixel (in linear mode) or 100 cents per pixel (in log mode) and lets you specify a different increment only if you already have an analysis open and a region selected. It has three additional options that determine how individual lines of pixels generate new frequency components: Pure Tones, which uses a 1-pixel-per-partial mapping and tends to produce rather static, harsh additive sounds; Random Noise Bands, which adds a bit of randomness to the partials' frequencies; and Track Frequency Spectrum, which is not clearly explained but produced the most interesting results in my tests regardless of the source files (see **Web Clip 1**).

One of the interesting options Audition provides is the ability to mask (filter) the spectrum of one file with that of another. For example, you could save the display of one sound's spectrum as a BMP file, then analyze a second sound and view its spectrum in the Spectral Frequency Display. You could then import the graphic image of the first sound's spectrum and use it to filter out some of the components of the second sound. In most cases, a good bit of trial and error is required, but the technique has a lot of potential for things like cross-synthesis and cloning the resonant qualities of one sound onto another.

Once you've got a graphic image converted into a spectral display, you can use Audition's powerful spectral-editing tools to modify the spectrum before resynthesizing it. For instance, you can isolate a small segment of the spectrum—all frequencies from 500 to 1,500 Hz between 3 and 6 seconds, for example—and then process just that region with reverb, EQ, or any other effect. And of course, with its robust multitrack mixing and editing options, Audition will make your newly generated audio file feel right at home when combined with any other audio files in your project.

Thomas Baudel's HighC 2.2 (Mac/Win, about $46 [MSRP])

HighC has strong ties to the original Xenakis UPIC system and in many ways is an enhancement to that system. The program's interface looks more like a traditional piano roll than the blank canvas found in most of the other programs, and many of the tools are optimized for drawing curves, lines, and gestures (see Fig. 3 and Web Clip 2). A typical session would start by using the Paint tool to draw some strokes on the canvas, picking a waveform and envelope to use for the sound file the image will generate, and then rendering the sound into audio. You can quickly build up echo, chorusing, or dense cluster effects by copying and pasting strokes, and easily adjust the total duration of the new sound regardless of the size of the original image you drew.

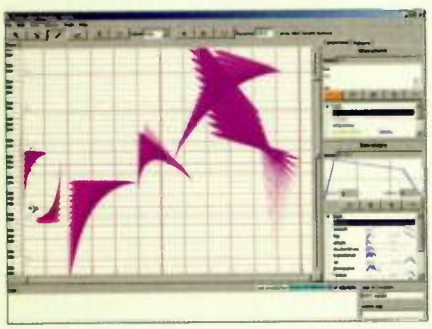


FIG. 3: This is the score for Rob Arnold's piece *Study After Xenakis* as shown in HighC. You can hear the resulting music in Web Clip 2.

You can create new waveforms for use in rendering, but depending on what type of waveforms you want, the process can be very simple or somewhat oblique. For instance, to build a static waveform containing only harmonic partials, you use an intuitive display in which the amplitude of each partial can be adjusted with a slider. Similarly, to create a noise-based waveform, you can adjust sliders representing the spread and density of the noise. For FM, however, you need to use the main graphic display and create an association between the source sound you want to modulate and a second sound that will serve as a modulator. I haven't seen this type of implementation in any prior software, but to be fair, it is modeled on the original UPIC approach (in fact, it offers much better visual feedback than UPIC), and with a bit of practice it becomes second nature. All new waveforms and any default waveforms that you modify are saved automatically when you save the piece you are working on, and you can access the waveforms you create in one piece when working in another.

HighC's drawing tools are fairly basic and though it currently uses only synthesis to create sounds, according to the developer a version that can also incorporate samples is in the works. Still, it is unique in its approach to working with image conversion and has a strong heritage from which many interesting compositions have been created. At the developer's Web site (see the online bonus material "Manufacturer Contacts"), you'll find a free trial version that doesn't allow you to export your audio file; links to lots of examples; and some handy tutorials to get you started.

Camel Audio's Cameleon 5000 1.5 (Mac/Win, $199 [MSRP])

Cameleon's image-to-sound conversion features are a relatively minor yet still very useful portion of the program. Like Audition, Cameleon treats audio files and images that you import in much the same way, but Cameleon, like Poseidon, provides an editable set of parameters for both types of files. You can use an image file to produce up to 64 harmonic partials and/or unlimited bands of noise, and you can adjust either the instantaneous amplitude of an individual partial or determine how it evolves over time using an intuitive interface for those purposes. You can also modify the frequencies of all or only some of the partials' frequencies using the Detune function, which could be useful, for example, for creating sounds with inharmonic spectra. Unlike some of the other programs, Cameleon doesn't handle stereo position (only the height and brightness of pixels are used), so color images will be interpreted as black and white.

Once you've imported a graphics file, you can make edits to selected groups of partials (only odd or only even partials, for instance), or use the Formant Filter (with its various presets) to sculpt out a portion of the sound. Cameleon also provides a number of preset spectra for both harmonic partials and noise components, so you could easily combine the spectrum generated by a graphic image with the noise components of a vocal preset and adjust the relative amounts of each.

You can use Cameleon's Morph Square to morph among four different sets of sound parameters that you've generated using up to four different graphics files, or mix and match sounds that you've created in different ways. It's easy to build your own pathways for automated morphing or use any of the Morph Timeline presets that come with the program. You can also adjust the Morph position in real time and capture that output to disc.

Cameleon will export a discrete time-domain plot (frequency versus amplitude) and convert it into a sonogram display. You can then manipulate this image with a graphics program or simply examine it to better understand the spectral components of your sounds. This technique, and others related to image-to-sound conversion, is nicely covered in a thorough text-based tutorial.

I wasn't very familiar with Cameleon before writing this roundup, but I found it to be one of the best-sounding soft synths I've come across in a long time. If you're looking for a professional soft synth that includes respectable image-to-sound features, Cameleon is a good place to start.

Rasmus Ekman's CoagulaLight 1.66 (Win, free)

CoagulaLight has been in "late beta" for some time but remains under active development. Though it won't replace Photoshop on your desktop, the program offers a wide range of brushes for creating new images as well as a rich set of tools for processing new or imported graphics files (see Fig. 4). Open an existing bitmap image, and you'll immediately have access to the Move, Zoom-rotate, and Skew-flip

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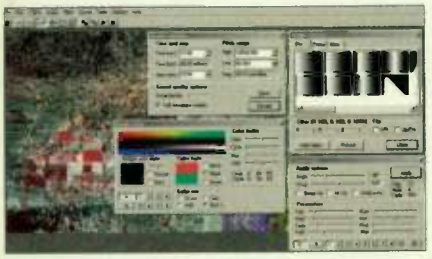

FIG. 4: Coagula has a robust set of brushes for drawing or modifying images.

features. Couple those tools with the ability to create mirror images in a single step, and you can quickly create complex mosaics and other highly transformed variations on your pictures before rendering them into sound.

Coagula lets you map the extracted image data to any arbitrary frequency range (between 0.001 Hz and any frequency) and adjust the new sound file's duration from a fraction of a second to several hours. You can enable the Soft Envelope Sweep feature to produce very smooth transitions between pixel values (with the trade-off of slightly longer render times), and you can choose to render only a selected portion of the image or stop the render midway (the portion that is rendered will be playable). A slider lets you dial in the amount of noise you want added to each sine wave individually before the final render.

If you're a fan of granular synthesis, you'll find the Spray brush very handy (note that Ekman is also the developer of Granulab, a very powerful, real-time granular-synthesis application). You can control the size and shape of the brush by clicking-and-dragging in the small window in the Brush dialog box—the brush will update as you move the mouse—and you can create color gradients that will translate into panning parameters of your new sound.

Like MetaSynth, Coagula provides several folders full of filters with which you can alter new or existing files, and you can adjust how much a filter impacts the R, G, and B values of an image independently. You can also add your own filters—any BMP file will do. I posted a collection of files for this purpose at emusician.com, some of which produce rhythmic patterns when applied to any image (see Web Clip 3). Try running several filters in series to produce polyrhythmic effects, or apply a filter using 100 percent R and 0 percent G, then flip and mirror the filter and apply it again using 100 percent G and 0 percent R to get different types of processing on the left and right channels (see Web Clips 4 and 5 for an example).

Nicolas Fournel's AudioPaint 2.1 (Win, free)

AudioPaint is an intuitive program that packs a lot of power under a very simple interface. It is one of only two programs here that can generate a new sound using image data to control samples, and it has a batch-processing option that allows you to generate dozens of individual audio files from a folder full of graphic images. Though it doesn't offer any painting tools or filters for editing an image you import or create

FIG. 5: This image was created using AudioPaint's Clouds of Points tool. Listen to Web Clip 6 to hear the sound this image generated.

in the program, the three image generators it provides are very flexible and tend to produce pictures that result in very interesting audio without a lot of effort by the user.

Clouds of Points, for instance, is used to create a grainy image with dozens or even hundreds of small colored points, each of which will trigger a sonic event (see Fig. 5 and Web Clip 6). You can determine the overall height and width of the image, the number of Clouds it will contain, and the number of points that will make up each Cloud. You can also pick the shape for the individual Clouds or simply choose the Random option, which lets the program pick all the parameter values randomly. Equally innovative and potentially useful is the Random Web Picture feature, which grabs nine random images from the Internet, displays them as thumbnails, then lets you pick the one that looks the most interesting.

As with the other programs, you can import a preexisting graphics file, but AudioPaint supports more formats than most of the rest of this group (JPEG, PNG, GIF, and BMP). Once an image is loaded, you can use the Audio Settings dialog box to pick the minimum and maximum frequencies for your new file (using hertz or note/octave increments) and choose between sine waves and a sample for the new sound's source. (Be sure to lower the maximum sampling rate if you're using samples, as the program does not filter out frequencies above the Nyquist frequency. Or perhaps you'd prefer to experiment with aliasing.) You can also determine which parameter of the image—whether Red, Green, Hue, Saturation, or Brightness—is used to generate the audio on the left and right channels independently.

AudioPaint may be a one-trick pony, but I was able to achieve very good results with fairly minimal effort, especially using the built-in generators. If you make it to Fournel's Web site, check out his other innovative freeware (all Windows only) to see if something else suits your fancy.

Image Line FL Studio 8 Express Edition (Win, $49)

FL Studio's BeepMap is a synthesis plug-in that first appeared in version 3.0 and that is available in all levels of the program, including the

FIG. 6: FL Studio's BeepMap will convert an image to any number of partials you specify. All of its controls operate in real time.

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entry-level Express Edition. It uses a graphic image (BMP, JPEG, or PNG) to generate a set of frequencies and amplitudes that control partials using additive resynthesis. In the conversion process, BeepMap assigns values extracted from red pixels to the left channel's amplitude and from green pixels to the right channel's. Yellow results in an equal value for both channels, and you can choose whether the program will ignore blue pixels or use their values to scale each pixel's frequency. There are only a few parameters to modify; for example, you can use one of three types of scales for the partials (logarithmic, linear, and harmonics). You can also determine the number of partials that will be used by specifying the maximum height (in pixel increments) for the image (see Fig. 6).

BeepMap has a Length parameter that will adjust the duration of your new sound from a fraction of a second to around 20 seconds (there are no values or increments shown for Length), and you can set the sound to loop or play back as a one-shot. Like other FL Studio generators, all of BeepMap's controls can be modified in real time. You can drag a new graphics file directly onto a button containing a BeepMap generator, even while a sequence is playing back, and integrate it into the current project. You can also send the output of a BeepMap generator to any FL Studio effect; a healthy amount of filtering and reverb can be effective, depending on the source image you're using.

U&I Software MetaSynth 4 (Mac, $499 [MSRP])

MetaSynth virtually defined the field of image-to-sound conversion on the modern desktop. It's long been the premier application in the field, and as it stands, it remains the most robust and

Izzy Gets Down

Mark Coniglio is the mastermind behind Isadora ($350), a real-time video- and audio-programming and performance environment that runs on both Mac and PC. Coniglio uses the software in his work as artistic codirector of the innovative New York–based dance company Troika Ranch (troikaranch.org) and has spent many hundreds of hours refining the system for real-time use.

Like Reaktor and other modular programming environments, Isadora offers a large number of video and audio modules (called Actors) that you connect on the main work area. A basic configuration might consist of a QuickTime movie running via the MoviePlayer, outputting to a video-processing module (a rectangular tiler, for example), which is then connected to the Projector Actor for desktop viewing. You could also enable output to an external device and capture it to a DV video camera or project it onto a large screen.

But things can get a lot more interesting really quickly. For instance, using the Sound Frequency and Sound Level Watchers, you could map the amplitude level of a user-defined frequency band of an incoming audio signal to one or more parameters of a video effect or image generator (see Fig. A). That way, the loudness of the incoming audio could control the amount of displacement in a displacement module or determine the number of particles in a particle animation created in real time by the program.

There are dozens of modules for mixing, generating, and processing both images and audio, as well as support for the FreeFrame plug-in standard (freeframe.org), so you can add additional commercial or free video-processing plug-ins, too. You'll also find math and I/O modules, and you can build your own interfaces using elements such as sliders, knobs, and dials. The best part is that you can freely mix modules of nearly any type, using data extracted from an audio signal as the input to a video-effect parameter or vice versa.

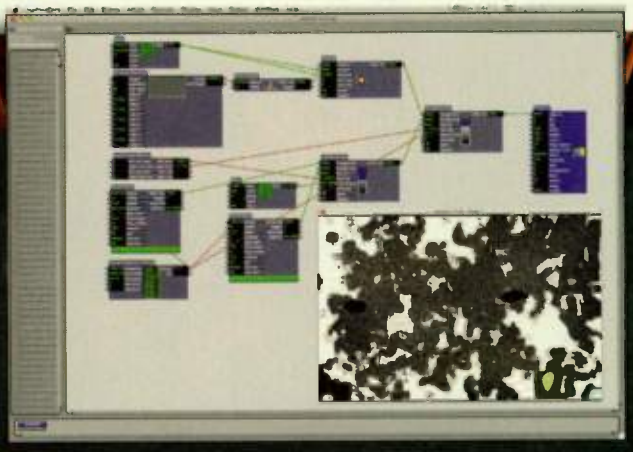

FIG. A: This image shows the main work area in Mark Coniglio's Isadora. The program's Actors (modules) appear on the left and are connected on the right. This configuration uses Actors to track the frequency and amplitude of an incoming sound source and map them to parameters controlling the amount of displacement that is applied to the source movie file. The final output is shown at the bottom right of the screen.

Even if you only want to work with audio modules, you can design a wide range of networks in which elements interact in innovative and unusual ways. For example, using the Core version of the program (which adds support for AU plug-ins), you could track the frequency output of one audio file, then use that information to determine the amount of delay or perhaps the pan position (or both) of a second sound. The same information, perhaps scaled by some factor, could also control another audio-effect parameter and of course a video effect simultaneously.

Isadora is supported by a terrific manual (PDF only), an active users forum, and a number of getting-started tutorials. If you want to explore combining video and audio or are simply looking for unique ways to interconnect audio parameters, then give it a try. A free trial version, as well as educational pricing ($275) on the commercial version, makes it well worth a look.

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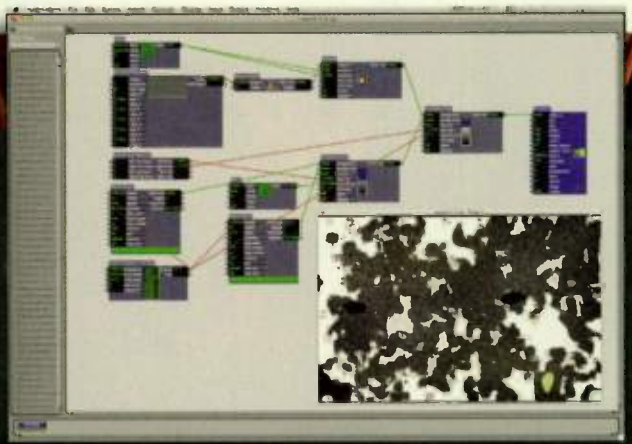

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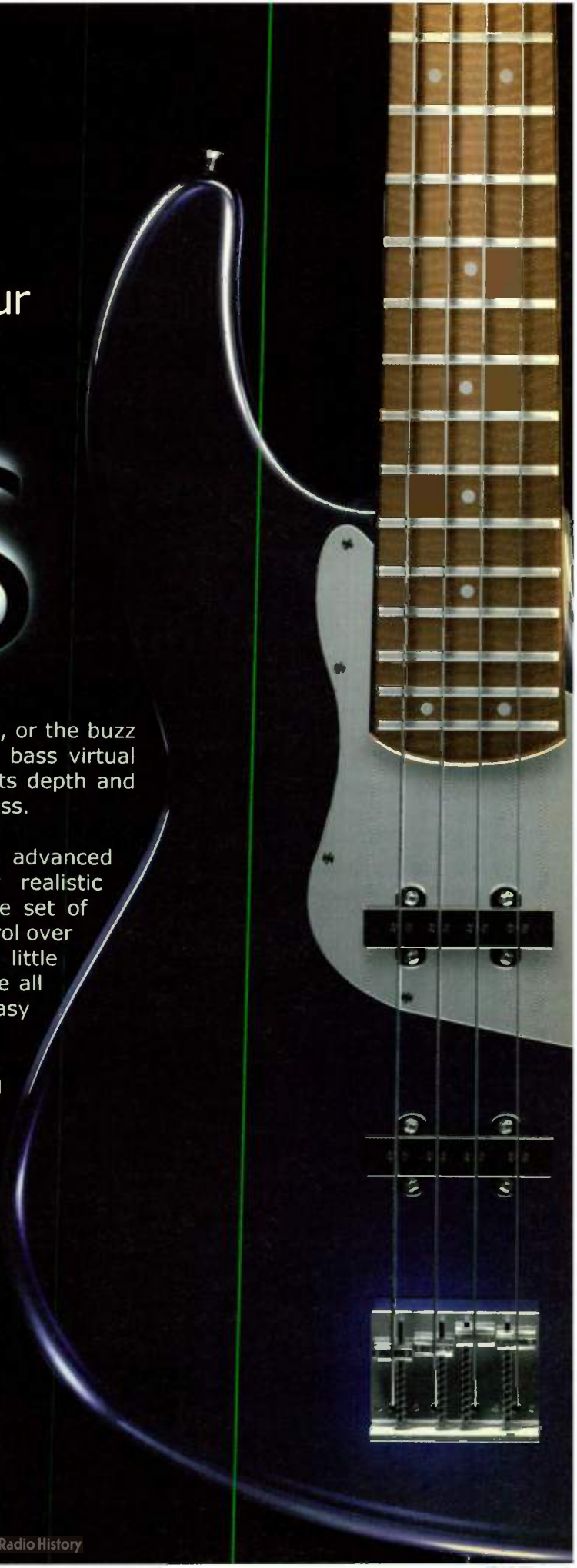

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versatile among this group. In addition to its graphics-related features, MetaSynth is a massive synthesis powerhouse, with tools for spectral analysis followed by multiple types of resynthesis, spectral morphing, sample processing, a variety of traditional synthesis techniques, effects, and much more. It even has a built-in timeline on which you can sequence sonic events.

MetaSynth is noted for its unique, rather dark and affected interface. But once you get past its exterior, you'll find an intuitive and well-thought-out structure. The program's features are grouped into six main work areas called Rooms. For image-related work, the two of most interest are the Image Synth and Image Filter Rooms. The Image Synth Room is where you load an image

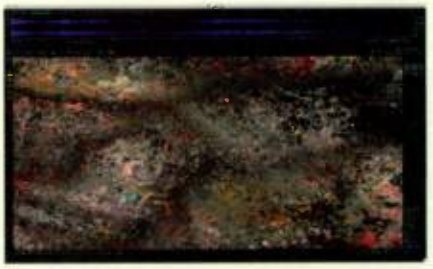

FIG. 7: It's easy to import an image file into MetaSynth and filter it, or you can draw a new image from scratch.

(PICT format only) and determine how it will be used to generate a new sound. (You can also draw a new image from scratch; see **Fig. 7** and **Web Clip 7**.) By default, the program will use additive synthesis, and you may find that many of your files initially produce similar-sounding results. But with only a few tweaks, you can switch to FM or granular synthesis, samples, or any arbitrary waveform you want, and that's just the first step of the conversion process.

MetaSynth lets you scale the duration of your new sound file up to a maximum of just under 13 minutes, and you can mix analyses of different source sounds. For example, you could analyze a graphic image, filter it with the spectrum of a vocal sound, then resynthesize it using a flute sample for each partial. The possibilities are limitless.

The Image Filter Room comes with dozens of preset filters, and it's simple to add your own. It's also easy to make pseudovocoder sounds or impose complex polyrhythmic patterns onto a static source sample. Though MetaSynth doesn't work entirely in real time, it does offer full-fidelity real-time previewing on most modern machines. Using a fast computer, you'll find it very comfortable to explore and experiment with the program's numerous processing functions.

Since its inception, MetaSynth has been a Mac-only program. If you're on a PC and really want to check it out, I suggest you do what I did: buy a Mac and give it a try.

VirSyn Poseidon 1.4 (Mac/Win, $279 [MSRP])

Like others in the VirSyn family, Poseidon has a somewhat unusual interface, offering the majority of its controls on only a single screen. The image conversion process, called Analyze

FIG. 8: Poseidon lets you choose how you want to display the spectrum of the image you import. The 3-D waterfall view is shown here.

Bitmap, is the counterpart to the sound-analysis command, Analyze Sound. Both run from the same menu, and both produce a 2-D (or 3-D if you prefer) spectral plot (see Fig. 8). Though you can't edit the analysis directly, you can determine the way it will be used to generate a new sound. For instance, you can loop through only a small segment of time, adjust the playback speed, modify the number of partials the new sound will contain (from 1 to 512), and determine whether it will include inharmonic or only harmonic partials. You can also map a number of these controls to MIDI data, which allows you to "perform" the modifications in real time.

Poseidon, like Cameleon, doesn't create a new sound file directly from an image but uses the analysis data, regardless of its source, as the core of a synth patch. You can save the data in Poseidon's VRD file format, then freely substitute the spectral analysis that a preset patch uses for data generated by the image. So, for example, if you have a vocal patch that uses envelopes and other parameters characteristic of vocal sounds, you could substitute the analysis of an image for the underlying sound that the vocal patch uses, thereby applying the vocal characteristics to the image-generated sound.

Poseidon runs as both a standalone application and a VST plug-in, and like other VirSyn software, it includes only fairly basic documentation. You'll also need a Syncrosoft dongle just to try out the demo of the program. But like Cameleon, Poseidon is a very good-sounding professional soft synth. With dozens of sound-editing options, you should find more than enough ways to tweak your image conversions into very musical results regardless of what you start with. EM

Associate Editor Dennis Miller is a composer and animator. Check out his work at www.dennismiller.neu.edu.

Dream

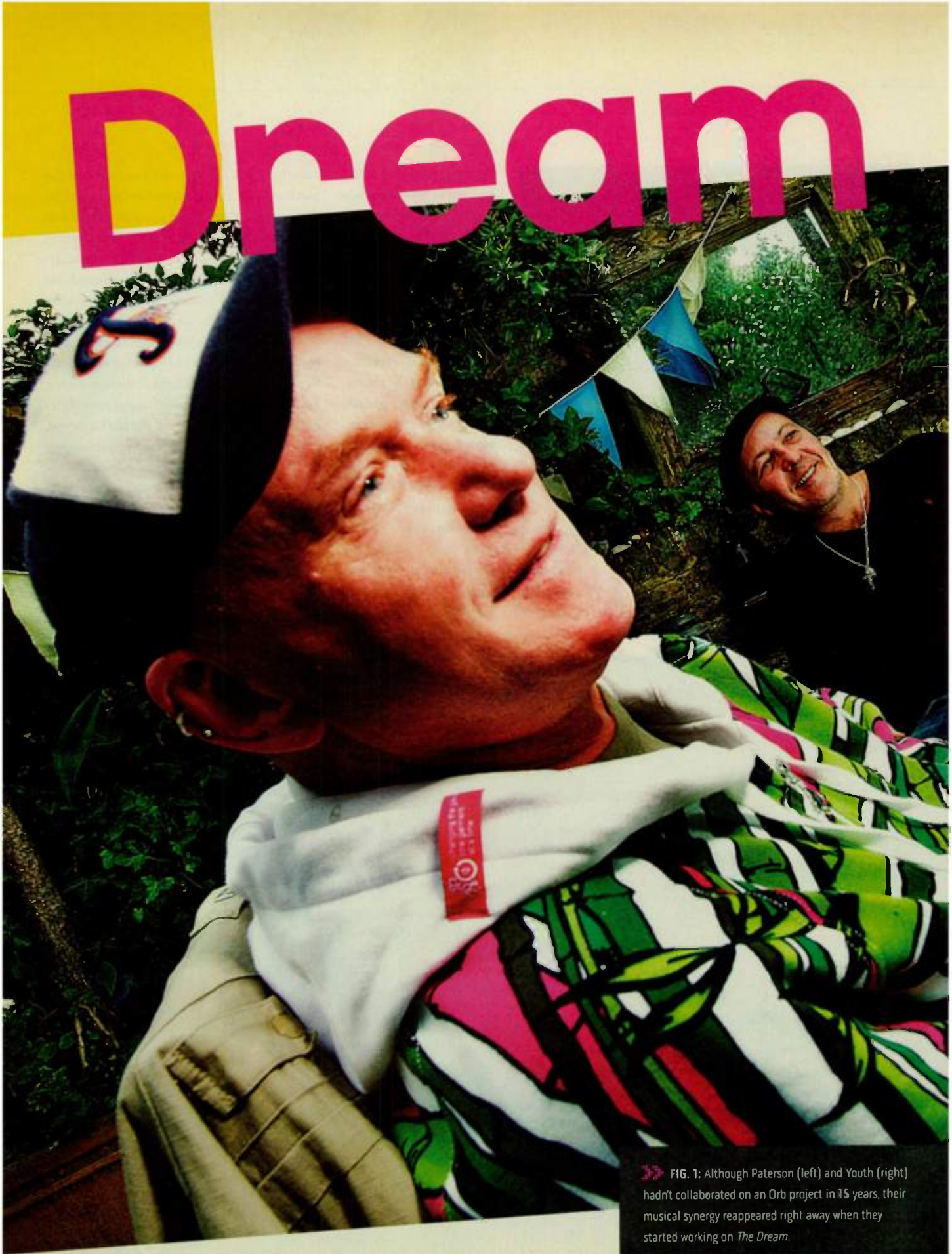

FIG. 1: Although Paterson (left) and Youth (right) hadn't collaborated on an Orb project in 15 years, their musical synergy reappeared right away when they started working on *The Dream*.

Producers

PHOTOGRAPHY BY CHRIS DAVISON

Alex Paterson and Martin "Youth" Glover join forces again on the Orb's latest ambient-dub exploration.

By Bill Murphy

The last time Alex Paterson and Martin "Youth" Glover worked in the studio together for any extended stretch of time was the early '90s. Yet when the two got back together at Youth's Dreaming Cave studio—along with engineer (and band member) Tim Bran and engineer-programmer David Nock—for the sessions that led to the Orb's new CD, *The Dream* (Six Degrees, 2008), the ideas started flowing almost immediately (see Fig. 1).

"I think the overall chemistry between me and Alex is pretty much as it ever was," Youth says. "We bring different things to the party now than we did then, but it's still a very natural process for us."

Paterson wryly admits that Youth's influence on *The Dream* resulted in a different-sounding Orb album than those of the recent past (see Fig. 2). "Youth would come in and wave his little production wand over everything and suggest things—like more vocals," he says with a chuckle. "I mean, if you've heard the *Okie Dokie* album [*Okie Dokie It's the Orb on Kompakt*, from 2005], there's not a vocal in there. And now suddenly we've got an album that's full of vocals and harmonies and choruses and bridges and things—it's kind of, 'Blimey, what's happening there?'"

Back in the Day

There was a time, nearly two decades and nine studio albums ago, when Paterson's mode of creation began and ended with three turntables, a bank of CD players, a few cassette decks, and an Akai 12-track mixer/recorder. As a DJ in the upstairs VIP chillout room at London's Heaven club, he was known for raiding everything from NASA space broadcasts to the ringing drone of Tibetan prayer bowls for his DJ sets—all of it grafted onto music by the likes of Brian Eno, Robert Fripp, Steve Hillage (who insisted on meeting Paterson the night he heard his own *Rainbow Dome Musick* album [Blue Plate, 1979] coming over

FIG. 2: One of the ways Youth influenced the production of *The Dream* was by convincing Paterson to bring in a number of vocalists, rather than making an instrumental record like the previous Orb release.

the speakers), Tangerine Dream, and 808 State. It was the beginning of a new genre called ambient house, and it was the future of the Orb.

The group started out as a duo. Paterson would tap into his vast library of samples, sound bites, and snippets, while his bandmate Jimi Cauty (then of acid house "situationists" the KLF) used an Akai S900 sampler, an Oberheim OB-8 synthesizer, and his studio smarts to massage the whole into what were often beatless, dreamlike soundscapes. But it wasn't until a disagreement between the two sent them on their separate ways that the Orb really took off.

Paterson had been childhood friends with Youth, who had made a splash of his own in the early '80s as the bassist with the postpunk outfit Killing Joke before moving on to become a stellar producer in his own right. Heavily influenced by the low-end sound of dub reggae (which was also near and dear to Paterson's heart), Youth hooked up with Paterson to record "Little Fluffy Clouds"—a 1990 sampladelic club hit that put the Orb on the map.

It soon led to a full-length album (1991's *The Orb's Adventures Beyond the Ultraworld*), but by the time the classic follow-up *U.F.Orb* had vaulted to No. 1 in the U.K., Youth's other gigs had pushed him out of the fold. Undeterred, Paterson soldiered on with Kris "Thrash" Weston, who had engineered most of *Ultraworld* and knew his way around a 24-track studio (as well as vintage Moog, VCS3, and Prophet-5 synths).

Preparing for Liftoff

For the sessions that produced *The Dream*, Paterson, Youth, Bran, and Nock would meet at the Dreaming Cave (see the sidebar "Fertile Ground" and Fig. 3)—usually on a very tight schedule—and would often set up to record almost as a live band would, and track what were essentially jams. As a song began to take shape, any one of the three bandmates might work on seeing it through to the next phase, depending on whose ideas won the vote.

"We'd just throw down some jams," recalls Youth. "We'd get a live performance and live dynamic going, and then a lot of it was arranged and edited later in [Apple] Logic [Pro]."

"Obviously there might be guest vocalists there, or other programmers helping out," Bran says, "but the core of the music was Youth playing bass, myself on keyboards, programming, synths, and samplers, and Alex providing the soundscapes, beats, and samples. We all found that way of working very exciting. We did it in very short, intense bursts, however—two or three hours at a time, because we were all doing other stuff. So from 10 in the morning until 1 in the afternoon, we'd just have a burst of creative energy, and then come back to it the next day."

> "You'd have cats walking through and children knocking on the door."

Can You Gear Me?

Each member of the Orb had a primary setup during the sessions for *The Dream*. For Youth, that was a vintage 1968 Gibson EB-2 short-scale bass, run through a DI box into the back of a TL Audio Tubetracker mixer for some added warmth before going into Logic. He often detunes the low E string to a D—as heard on the bottom-heavy "Dirty Disco Dub"—and will play the entire bass line on that one string to get a consistent tone, in the manner of Jamaican bass legends like Family Man Barrett and Robbie Shakespeare.

"If I'm on the Logic [Pro] EQ," Youth says, "I'll just slightly dip the bass and push the midtones up to brighten it up a little bit, just to give it a little bit more cut, but not much. It depends. There are certain parts where I might go for a really middly, twangy, reverby sound, and then for other bits I'll just have the normal EB sound. I don't really need to EQ it much because it's quite boomy and round and smooth, with this really pure, deep tone that sustains forever. I just ride that tone and try to be careful not to wobble my finger or use any kind of subtle intonation because that will throw it out."

As the deviant genius with the massive record collection and the penchant for recording everything he hears or watches on TV, Paterson crafts the sample-based atmospherics that propel any Orb project. For tracking sessions on *The Dream*, he would bring in a number of audio sources (vinyl, CDs, and even old mix tapes on cassette), which he and Bran would load into Ableton Live and then chop, loop, and arrange. From there, he used a stripped-down DJ setup—an old belt-drive turntable, a Pioneer CDJ-1000 MK2 digital deck, a Korg Kaoss Pad 3, and a basic 2-channel mixer—for his live-in-the-studio takes.

"It was very limited for what I had," Paterson admits, "but then what we had with the actual software within the computer—that was limitless, what we could do." Another crucial compositional tool from a beat perspective was Paterson's Korg Electribe ER-1 drum machine, which would be MIDI synced to lock it to a groove and then "played" during a jam, much like any other instrument.

"It's a great thing for actually getting your hands on and changing the filters," Bran says. "I remember me and Alex putting down a few parts and both jumping on the buttons to get a few rhythms and jam along, basically. A lot

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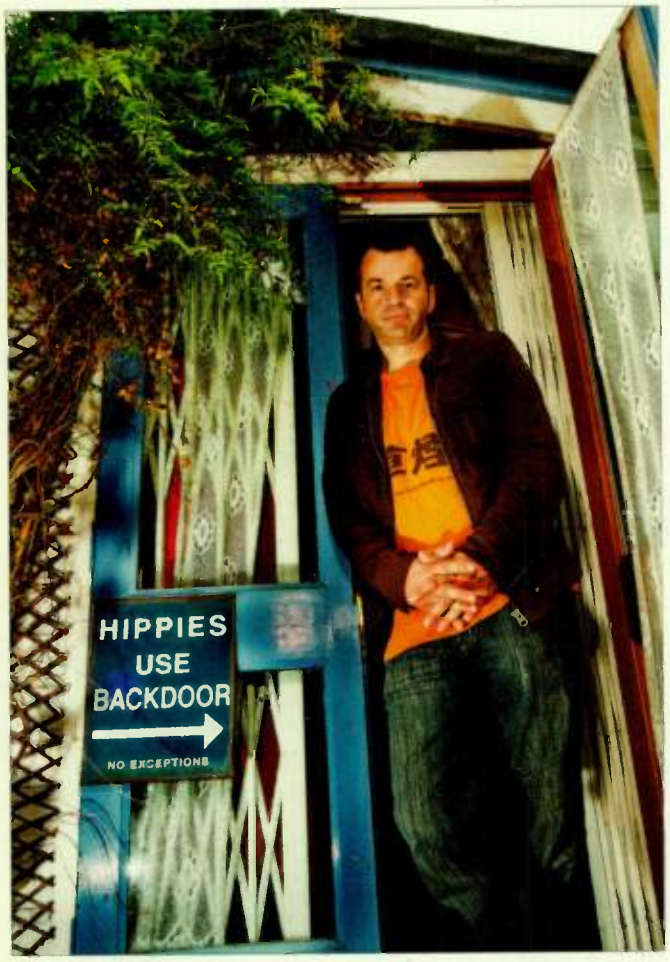

FIG. 3: Tim Bran, shown here standing outside the Dreaming Cave, contributed both keyboard playing and engineering on the project.

of the songs came from just literally getting a beat on the Electribe and me on keyboards and Youth on bass and maybe a singer there sometimes live, and just having a jam and then chopping that about. We'd find the hooks and then build on that."

Bran would often jump on one of the many analog synths that populate the Dreaming Cave—including a Roland Juno 106 and a vintage Korg MS-20—but his instrument of choice is a Clavia Nord Lead 2. "Alex would play a record and say, 'I like that kind of sound,' and I could pretty much copy the sonic quality of it in the Nord," he explains. "It gives you that real edgy, spiky digital sound, but you can also really get some nice analog sweeps. I know it really well, so I know straight where to go to get the sounds."

Dub Gone Crazy

As fans of the deep-pocket psychedelic mixing style known as "dub" pioneered in the early '70s by Jamaican record producers like King Tubby and Lee Perry, Paterson and Youth were on the same wavelength about how low end and freaky atmospherics would interact and mesh together on *The Dream*. The leadoff title track is a prime example: opening with the distantly flanged guitars of Matt Chandler burbling in the background, the song quickly gives way to synth swells, disembodied voices, vinyl scratches, and nature sounds squeezed through all manner of tape delay and reverb (from Logic Pro's native effects suite) before the bass and beat come in thick.

"We often put the bass through filters in the Korg MS-20 and I'll modulate it manually," Youth says. He also favors Line 6 stompboxes and Camel Audio plug-ins for various effects. "We'll have a long delay on it that comes back through the desk, and then we'll filter the return of that, and send that to a reverb, all while I'm modulating the MS-20. So you get this four-levels-of-filter-delay thing going on, which can be really good fun."

Although *The Dream* is front-loaded with a string of vocal-based songs, the dub thread runs through most of them, emerging full tilt as the album turns progressively darker and weirder. The highlight is "High Noon"—a Shaolin-meets-spaghetti-western trip to the outback that's rendered complete with the twangy guitar lines of Steve Hillage (working the virtual knobs of Native Instruments' Guitar Rig plug-in to their logical extremes). The song cycles through aggressively signal-processed soundscapes that recall vintage Orb circa 1993, tempered with a liberal dose of the electronic dub style à la Massive Attack or, to go truly old school, Mad Professor and Jah Shaka. As Paterson explains it, the adventurous sonic explorations of dub music are in his blood.

"It's been bred into me from growing up with it, really," he says. "I know the dub sound with my eyes closed—one ear closed, even. It's all the Joe Gibbs style, and King Tubby's style, Scientist, and even Adrian Sherwood. Those are dub engineers, and that's where we really did take it on probably the first six albums of the Orb."

Voices Carry

Unlike many of the recent Orb albums Paterson recorded with Berlin-based electronics guru Thomas Fehlmann, *The Dream* flows with a prominent vocal presence from several guest singers, including Aki Omori (on the album's current single "Vuja De"), Helen Boulding ("The Dream"), Juliet Roberts ("A Beautiful Day" and "Mother Nature"), and upstart ragga toaster the Corpral ("Dirty Disco Dub" and "Lost & Found"), who has joined the live Orb lineup on tour this year. Paterson finds it a bit unusual to be so reliant on vocals for an Orb project, but he stresses the importance of how they serve the song—particularly when a sampled passage has to be redone for legal reasons.

"We learned our lesson straightaway [back in 1989] with 'Huge Ever-Growing Pulsating Brain,' which used a Minnie Riperton sample," Paterson says, referring to one of his earliest singles, which ended up on 1991's *Ultraworld* album. "You can't do that—it's naughty. You've either got to get it resung or replayed or do something original, and this is what we did for the vocal line on 'Vuja De.' I got Aki to redo a Liz Torres–style, 'You Used to Hold Me' feel—that's where the vocal elements developed from." ("You Used to Hold Me" was a white-label B-side to Liz Torres's 1987 12-inch "Can't Get Enough" and was actually sung by Xaviera Gold.)

Short of singing in the shower, tracking vocals at the Dreaming Cave is probably about as laid-back and relaxed as it can get. "We've just got a little vocal booth at the studio," Bran explains. "It's literally just a wooden door with planks in it, and it's not soundproofed in any way. You'd have cats walking through and children knocking on

the door to come and see their dad *[laughs]*. But I think it actually puts less pressure on a vocalist to have that sometimes. You don't have a talkback mic—you just talk through the door—so they almost feel like they're in the same room."

Vocals were usually recorded using a Neumann U87 microphone and then processed according to the demands of the song. For Juliet Roberts' catchy hook on "A Beautiful Day," for example, a bit of compression helped her layered harmonies sit snugly in the mix. "We put her through a real [Universal Audio] 1176," Bran recalls, "and then maybe through a bit of Focusrite EQ, and then into the Fireface and into Logic. And that was literally just one take. She would be hearing what we heard on the [live] mix and that would be it—no cue mixes or anything. It was very simple and straightforward."

The Orb: A Selected Discography

Okie Dokie It's the Orb on Kompakt **(Kompakt, 2005)**
It's been speculated that Thomas Fehlmann asserts more of his presence on this album because he was the go-between for the Orb's deal with the Kompakt label. While it's true that the lengthy ambient odysseys of earlier Orb albums are largely absent here, Paterson still throws some beautiful wrenches into the works, particularly on the album's sample-fractured closer, "Snowbow."

Cydonia **(Island, 2001)**
Even with its release date delayed by two years, *Cydonia* was perhaps a little ahead of (or out of) its time, given the critical drubbing it took in the British press. No doubt the group's decision to go with a more song-oriented approach—complete with guest vocalists Aki Omori and Nina Walsh on several tracks—contributed to the discontent, but there are still some classic Orb moments here, including the hypnotic flute runs and hyperprocessed drums of "Promis" and the Tangerine Dream–like synth arpeggios that open the dark epic "A Mile Long Lump of Lard."

Orblivion **(Island, 1997)**
Honed down to the production trio of Paterson, Andy Hughes, and Fehlmann, the Orb here delivers a brighter, tightly orchestrated, beat-heavy sound that's by turns mystical ("Ubiquity" and "Bedouin") and made for the dance floor ("Asylum"). The album also includes the notorious single "Toxygene," which was originally commissioned—and summarily rejected—by French synth-wave godfather Jean Michel Jarre as a remix of his trance sleeper "Oxygene 8."

U.F.Orb **(Big Life, 1992)**
Propelled by the cerebral space-bleep epic "Blue Room"—which features Jah Wobble on bass and Steve Hillage on guitar, and an unheard-of maxisingle form that clocks in at around 40 minutes—this richly recorded suite of fat synth excursions is now an ambient classic. It was also marked by Youth's last contribution to the Orb canon (the driving, Autobahn-esque "Majestic") and found Paterson biting off samples of spoken dialog with Monty Python–like whimsy ("Towers of Dub").

The Orb's Adventures Beyond the Ultraworld **(Big Life, 1991)**
On the strength of "Little Fluffy Clouds" alone—instantly memorable for its snippet of Rickie Lee Jones recalling the beautiful skies of her Arizona childhood—*Ultraworld* is a sample-crazy masterpiece, but it's also a double-disc triumph of collaboration. Tracked in no less than six different studios with more than a dozen outside musicians (including Trevor Horn, Hillage, and session bassist Guy Pratt), it's a sprawling odyssey that somehow still manages to sound like a seamless whole.

>> FIG. A: Inside the Dreaming Cave studio, where Youth, Paterson, and Bran would set up and play live, capturing jams that were used as the basic tracks for *The Dream*.

Fertile Ground

Located on the near-idyllic grounds of Youth's home near Wandsworth Common in South London, the Dreaming Cave occupies a wooden summer shed at the end of a long, narrow garden—the perfect private getaway for tracking a concept-heavy album like *The Dream*. Fittingly, Youth likens the studio to the TARDIS time machine from the popular *Doctor Who* sci-fi series: it seems bigger on the inside than it looks from the outside (see **Fig. A**).

"I can get a band in there with a drum kit," he explains, "and I have a small TL Audio Tubetracker [M4] desk as well. It looks tiny from the outside, and then you go in and it's full of old analog stuff and a big Pro Tools/Logic rig. And I actually just replaced the Pro Tools rig with an RME Fireface 800 and kept the Logic native setup when we got into Logic Pro 7 [now version 7.2]. I found that to be faster and better than working on Pro Tools."

Youth points out that he still keeps a mobile Pro Tools HD rig for recording live shows, but in the end it was the ease of working with Logic that sold him. But Bran, who has worked with Youth since 2004 on numerous projects, including the Orb, says that due to the difference in plug-in formats between Pro Tools and Logic Pro, the transition wasn't totally seamless.

"We're writing and mixing as we go with the Orb," Bran says, "so a lot of the mixing tends to get done inside the box. I remember that when Youth decided to sell his [Pro Tools] TDM rig, we'd already mixed about half of the album. And I suddenly woke up one morning and thought, 'Hold it—some of our in-the-box mixes aren't gonna work. I've only got a day to transfer everything to Core Audio.' I think I even did stems, just to get everything bounced. But it was a good change. We've got a Mac G5 now, and it's even more instant and accessible when you've got the Core Audio [Audio Units] plug-ins, I think. TDM is great, and you can get some really professional sounds, but sometimes you want the mad VST stuff—the one plug-in that some guy has written that you can't get anywhere else." Bran used Ableton Live as a host when he wanted to use VST plug-ins on the project.

The Dreaming Cave also has a Universal Audio Ultra Pak DSP card installed on the studio's main computer—a 2.5 GHz Apple Mac G5 Quad. Youth's and Bran's favored plug-ins include MOTU's Ethno (used liberally on "Mother Nature," with Steve Hillage), PSP's Vintage-Warmer 2, GForce's Oddity and impOSCar synth emulators, and Smartelectronix LiveCut for simulating turntable scratches.

Bran is also a frequent flier at KVR Audio (kvraudio.com), a searchable online database for VST and AU plug-ins. Some of his recent favorites: TAL Dub Delay, Big Tick Cheeze Machine ("a really nice string synth plug-in, especially if you like old-sounding ARP Solina strings"), a TB-303 emulator by Muon Software, and the reFX Vanguard soft synth.

Director's Cut

Although virtually all of *The Dream* was premixed within Logic Pro—after getting plenty of analog warmth on the front end, whether on Youth's bass, Bran's synths, or the various lead vocals—some of the final mixes were farmed out to a couple of key personalities from past Orb transmissions. Greg Hunter (an off-and-on Orb traveler since 1990) puts his stamp on "Katskills" and "Mother Nature," while Andy Hughes (formerly a permanent member of the group, beginning with *Live 93* [Island, 1993] and solidified on

> "We did it in very short, intense bursts, however—two or three hours at a time."

Orbus Terrarum [Island, 1995]) lends his signature to "Vuja De" and "Dirty Disco Dub."

Bran points out that he, Youth, and Paterson were happy with one of their own final mixes of "Vuja De" but that Hughes extended, and thus profoundly energized, the song even further. "He added that whole ambient tail-out," he says. "There's a big long analog chord that holds, and then the track fades out. There were a couple of other ideas I think he changed from the original mix, but basically he ended up sounding more commercial with his vision on it, and I really love it. I know he uses a TDM system—maybe Logic over a [Digidesign] 888—so he would have had different plug-ins from what we used. He certainly gave those two mixes a lot more clarity and clout."

Of course, with low end being such an essential ingredient in the classic Orb sound, special attention has to be paid to the elements of the rhythm section when it comes time to mix. "It can be tricky with the Orb," Youth says, "because there's so much sub coming through the synths and drum programming that you have to have it fitting nicely. You're always compromising the bass guitar with the sub of the synths or the drums, so it's just a matter of balancing it out, I think. Generally, if there's a bit of a clash, I'll adjust my playing [from the outset], or we might EQ my bass later on to compensate."

For well over a year now, Paterson has been involved in preparing the entire Orb catalog for a massive reissue (with bonus discs, new sleeve notes, and all the bells and whistles), even as he continues to keep his hand in mix projects such as *I'll Be Black* (2007) for the legendary Trojan label. He still DJ's regularly and is looking ahead to more live gigs with the Orb in the United States this fall—particularly if he can get Youth, who will be touring simultaneously with a reunited Killing Joke—in on the act. When asked if *The Dream* is an ambient throwback, of sorts, to the vintage Orb sound, he waxes downright philosophical.

"I think it's almost like a dream of the past," he says. "Maybe this is how an Orb album turned out in the future, if it was an alternative future. Maybe we were doing things a little bit too early for people to get their heads around in the mid-'90s. Now I'm 48, so it's getting kind of difficult to prophesize things that really should just take a natural turn musically. Hopefully, if I can get more people going through my music, they can turn that corner themselves and make music that's not that far down the road from the Orb. Anyway, there you go. That's the ambient answer: who knows?"

ONLINE BONUS MATERIAL

(Editor's note: Check out the online bonus material at emusician.com for supplemental audio interviews with the Orb.) EM

Bill Murphy is a regular features contributor to Remix *and writes online content for various music-software developers. From 1996 to 2002, he ran the Axiom label for producer Bill Laswell.*

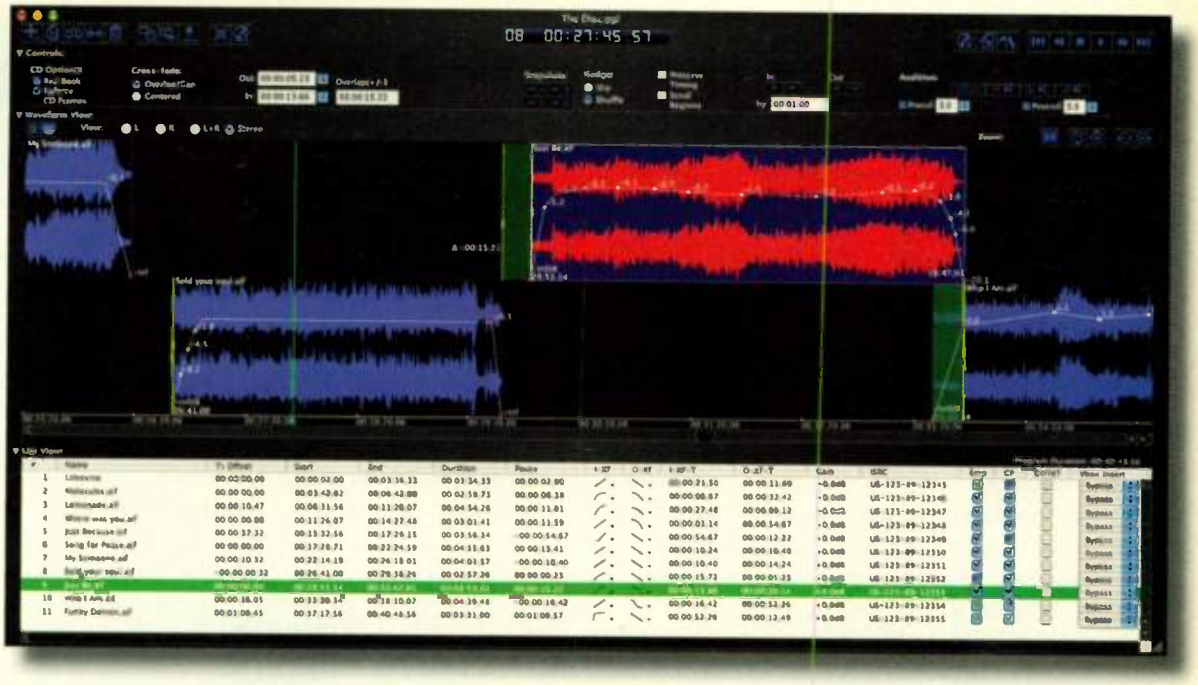

FIG. 1: In BIAS Peak Pro 6's Playlist, the control area is at the top, with the Waveform View in the center and the List View on the bottom.

Mastering with Peak Performance

In BIAS Peak Pro 6, a little exploration yields big rewards.

By Larry the O

BIAS Peak Pro 6 is an extremely feature-rich environment for the Mac that's used to record, edit, process, and deliver audio. This article will focus on its mastering application, for which the program has been widely used during most of its existence.

A Room with Two Views

Although files are often edited and processed as individual documents before being assembled into the final work, mastering in Peak Pro 6 occurs primarily in the program's Playlist environment (see Fig. 1). The Playlist, which was introduced in Peak 2, has received a major upgrade in Peak Pro 6.

The Playlist window has three main areas: the Controls section at the top; the Waveform View section, which provides a graphical editing interface; and the List View, which has a text-based interface. You can perform many functions in both the Waveform View and the List View that enable you to choose between the intuitiveness of graphical editing and the precision of entering values directly.

Each view also has capabilities that are not available in the other view. The Waveform View makes it easier to customize crossfades, perform volume automation, and skip around during auditioning, whereas the List View offers the ability to configure subcode flags, enter metadata such as CD-TEXT and ISRC codes, and add processing to each Playlist Event.

Crossfades: Heart of the Matter

Creating and shaping crossfades is a primary mastering task, and Peak Pro 6 supplies a large array of features for crafting them to fit individual needs. The Playlist has two fundamental crossfade modes, Overlap and Centered, and two primary tools, Move Event/Crossfade and Trim Event (see Fig. 2). The Move Event tool appears only in Overlap mode, and the Crossfade tool takes its place in Centered mode. The actions of these tools are modified

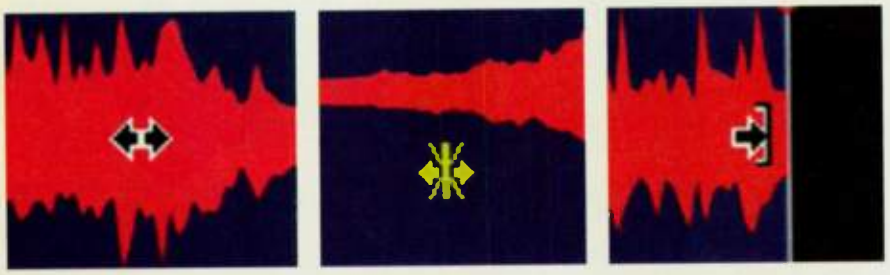

FIG. 2: The Move Event tool (left) is available when working in Overlap mode, whereas the Crossfade tool (center) replaces the Move Event tool in Centered mode. The Trim Event tool (right) is available in both Overlap and Centered modes.

by the state of the Preserve Timing and Scroll Regions checkboxes.

Overlap crossfades—the kind most commonly found in DAWs and other editing programs—are used for sequencing takes or assembly editing. By default, you can adjust the crossfade's fade-out component (from the earlier Event) and fade-in component (from the later Event) independently of each other. Start and end times of one component aren't necessarily related to those of the other component, and there can be any amount of overlap between the Events, including a negative overlap (a gap between Events). The Events themselves may get moved in time while constructing an overlap crossfade.

Centered crossfades are mostly used for digital splicing, such as compositing multiple

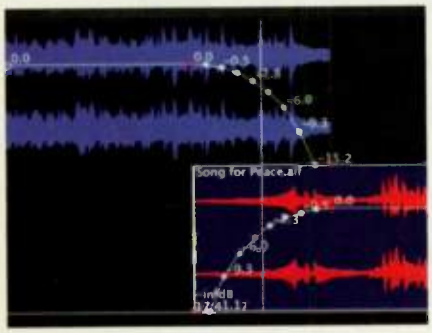

FIG. 3: In centered crossfades such as the one shown here, the crossfade establishes an edit point based on the time relationship between the two Events. The edit point is always at the midpoint of each component's duration and will remain fixed as you edit the crossfade.

takes together (see Fig. 3). Centered crossfades are often used in editing classical music, where a large number of splices are used in a single movement of a piece. A centered crossfade is focused around an edit point that's always anchored at the crossfade's midpoint. When you adjust a centered crossfade, the edit point and the temporal relationship between the two Playlist Events remain fixed, and the start and end times change symmetrically on either side of the edit center point. Peak Pro 6's Nudge buttons enable you to precisely adjust the crossfade parameters. (For more on crossfade techniques, see the online bonus material at emusician.com.)

Fade Shapes

In the Playlist window's Waveform View, you can customize fade shapes in any desired fashion with simple breakpoint editing. The Zoom To Fit Transition button, which is situated next to the zoom buttons immediately above the Waveform View's right side, gets you up close to the crossfade for more-detailed editing.

In many situations, it is desirable to use the same shape for all fades. The Playlist allows for designating any available envelope shape as the default fade shape. The Playlist Default Fades dialog box appears whenever a new Playlist is created, but you can invoke it at any time by clicking on the Set Default Fades For Playlist button in the control area's upper left portion (see Fig. 4).

Clicking on the checkbox that's labeled Automatically Apply Default Settings When Adding Events To Playlist will cause the fade shapes you've selected in this dialog box to be applied each time you add a region as a new Playlist Event. To apply the default shape to an existing Event, choose Default from the drop-down menu for the desired fade in the List View.

You can further customize the fade shapes with breakpoint editing. In the Playlist Default Fades dialog box, click on the In Shape or Out Shape box beneath the fade-duration field to open Peak Pro 6's envelope editor. The pull-down menu at the top of the dialog box will show all envelope shapes present in the Peak Envelopes folder of the user's Library →Preferences folder. You can create and save custom fade shapes in the Crossfade In or Crossfade Out editor.

With all of these manipulation capabilities, it is essential to have flexible facilities for auditioning crossfades. The control area's Audition section contains buttons that allow you to audition either the fade-in or fade-out component or the entire transition, with programmable pre- and postroll.

Adding Your Special Sauce

In virtually all cases, mastering entails adding EQ, dynamics, reverberation, or other processing to one or more tracks. It is common to have some processing that is applied to all tracks.

ONLINE BONUS MATERIAL

To process any individual Event in the Playlist's List View, choose the Active command from the Vbox Insert field's drop-down menu for that Event. Vbox will then appear, allowing you to insert one or more VST or AU plug-ins.

Peak Pro 6 XT includes BIAS's Master Perfection Suite (MPS), which contains high-quality equalization, linear-phase multiband compression, gating, spectral cloning, and analysis and metering tools, which cover the mastering processes that most users need. The XT package also includes SoundSoap Pro to tackle audio-restoration tasks.

Once configured, choosing Make Snapshot

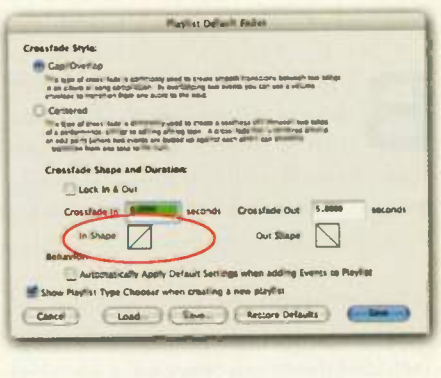

FIG. 4: The Playlist Default Fades dialog box lets you configure fade shapes and durations that can be applied to every new Event. Clicking on either the Out Shape or the In Shape box (circled) opens an envelope editor for creating custom fade shapes.

"I've Already Earned $50,000 Using TAXI and My Little Home Studio."

Matt Hirt – TAXI Member

Is your music good enough to make money?

I was pretty sure mine was too, but I didn't have a clue how to make great connections. I'm just not good at playing the "schmoozing" game. And even if I was, I had little chance of meeting the right people.

I needed a way to market my music, so I joined TAXI and the results were nothing short of incredible.

Now, all I have to worry about is making great music. The people at TAXI do an amazing job of hooking me up with opportunities that I would never uncover on my own.

I've already cut deals for more than 70 of my songs, and they're getting used in TV shows like Dateline, Law and Order SVU, and The Osbournes. And yes, I'm making money.

I was kind of surprised that the recordings I make in my little home studio were good enough. I guess size really doesn't matter;-)

Want to know what does matter? Versatility. Being able to supply tracks in different genres makes you even more desirable for Film and TV projects. I didn't know that until I became a TAXI member and started going to their members-only convention, the Road Rally.

If you joined TAXI and never sent in a single song, you'd still get more than your money's worth just by going to their convention. It's three days of incredible panels loaded with some of the most powerful people in the music business, and the cool part is that it's FREE!

Unlike some of the other conventions I've attended, the panelists at the Rally are friendly and accessible. I've never been anywhere that gives you so much great information, and so many chances to meet people who can help your career.

If you've needed proof that a regular guy with ordinary equipment can be successful at placing music in TV shows and movies, then my story should do the trick.

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from the Vbox drop-down menu will create a snapshot of that Event's processing configuration. You can easily copy the processing chain to another Event by choosing Make Snapshot from the drop-down menu in the target Event's Vbox field.

Master processing is accomplished using the five inserts in the main application's Plug-Ins menu. The standard Peak Pro 6 package includes Reveal LE—a special, Peak-only version of MPS's metering and analysis suite. Instantiating Reveal LE in one of the master inserts lets you see the exact level, frequency response, and phase characteristics of your audio.

Special Delivery

As a mastering tool, Peak Pro 6 provides a variety of options for the final delivery format; three buttons in the Playlist window's upper right portion open the door to the available choices. The leftmost button burns the selected Events to an audio CD. To produce an output that's properly formatted to the Red Book standard, make certain that the Red Book and Enforce CD Frames boxes are checked before you start assembling your Playlist. Only in a few rare, highly specialized applications is it desirable to have Enforce CD Frames unchecked.

You may be creating regions in an audio document (rather than in a Playlist) from which you intend to make a Red Book disc. If that's the case, choose the Action→Snap To→ CD Frames command to ensure that no clicks occur when CD frame boundaries are enforced in the Playlist.

Clicking on the Bounce Playlist To New Audio Document button (to the right of the Burn Playlist To Audio CD button) opens a dialog box that lets you specify the details for the file. Take a close look at the File Format drop-down menu, which shows the most commonly used formats for this application. Standard choices such as AIFF, Sound Designer II, and JAM Image files are there, but you can also choose to bounce to an iTunes Playlist.

If you're making masters for CD replication, you'll be particularly interested in DDP 2.0, a choice in the File Format drop-down menu. Disc Description Protocol (DDP) capability is standard in Peak Pro XT and optional with the regular Peak Pro package. DDP was created by Doug Carson Associates (which makes disc premanufacturing products) and is the format of choice for delivering CDs and DVDs for replication. Peak Pro 6's DDP 2.0 output can even include diacritical marks in the CD-TEXT data.

Any time that you convert the audio's sampling rate or bit depth to the format you'll need for final delivery, it's best to perform those operations while bouncing. Both functions are available in the Bounce Playlist dialog box. Peak Pro 6's sampling-rate conversion is used any time you select a target sampling rate that differs from the source rate; no other action is required to invoke this.

When you select a target bit depth that's less than the source material's bit depth, the Dither checkbox in the Bounce dialog box becomes active. Clicking on the Dither Prefs button will invoke BIAS's Dither Cloning Audio Technology, which offers an exceptionally wide selection of dither shapes, many of which are modeled on popular dither characteristics. Your choice of dither should be based on the nature of the source material and whether any further processing (such as conversion to MP3 or AAC) will be performed. Brief descriptions of each choice in the Dither Prefs dialog box can help guide you, but your ear should always be the final arbiter.

The rightmost of the three output buttons enables you to print or export a detailed text report of the Playlist. The dialog box that comes up when you click on the button lets you choose which data is included (see Fig. 5).

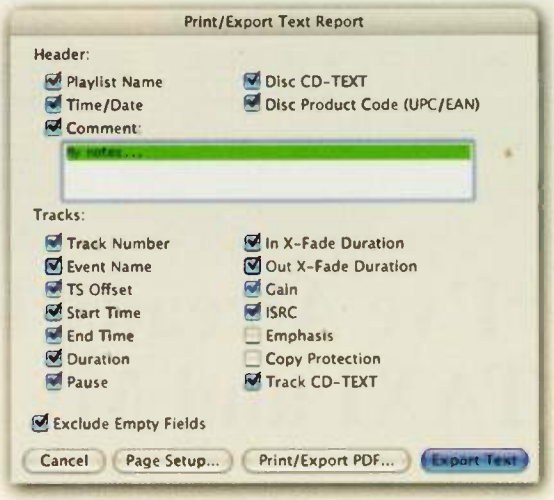

FIG. 5: The Print/Export Text Report dialog box enables detailed specification of the information that will be included in the formatted report that Peak Pro 6 generates.

> The Playlist received a major upgrade in Peak Pro 6.

Maximum Performance

Peak Pro 6 is used by sound designers, dialog editors, Podcast creators, and many others with various audio needs, but mastering has long been a primary focus of the program. Peak Pro 6 represents the culmination of years of providing mastering facilities, and it presents a rich feature set for mastering work. As with any powerful tool, however, taking time to explore the program and its documentation will reveal features that offer both convenience and the kinds of tweaks to fine details that are essential for mastering, which is possibly the most exacting of audio jobs. em

A longtime contributor to Electronic Musician, *Larry the O is a musician, composer, sound designer, engineer, producer, and general troublemaker in the San Francisco Bay Area.*

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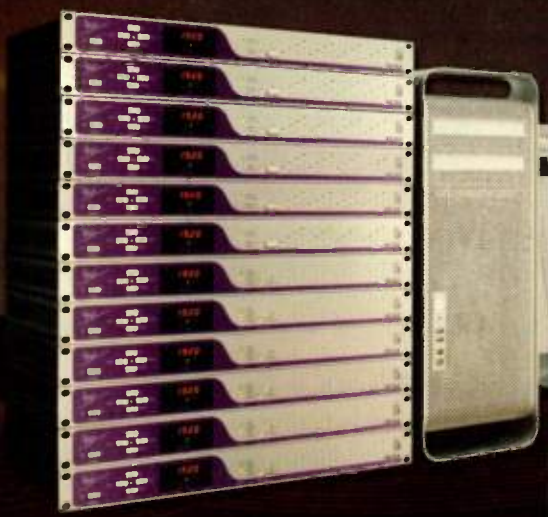

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Pitch Doctor

Modify melodies using Digital Performer's pitch-automation tools. | By Michael Cooper

In a me-too world where competing DAWs offer essentially the same features, MOTU Digital Performer (DP) separates itself from the pack with its standout pitch-automation functionality. The two most widely known applications for this type of processing are real-time pitch correction and conversion of audio pitches to MIDI data (used, for example, to double monophonic audio phrases on a virtual instrument).

Another little-known but invaluable application is nondestructive rearranging of musical parts in an audio track. Long after the guitarist has packed up and left the studio, for instance, you can change the notes he played to create a better hook—all in real time and without reaching for the record button. You do that by automating the pitch transposition of individual notes, and I'll show you how in this article (see "Step-by-Step Instructions").

What's on the Menu?

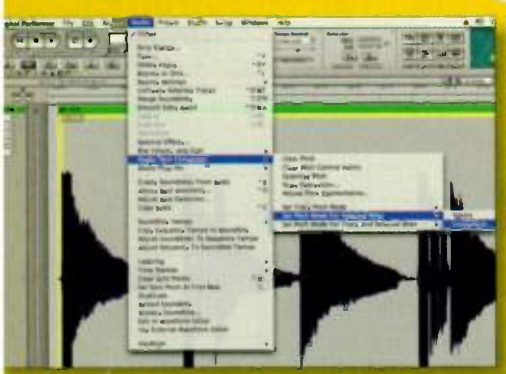
Before you begin, make sure PureDSP file analysis is enabled by navigating to Digital Performer→Preferences→Background Processing and selecting Analyze Soundfiles For DSP As Soon As Possible. Also activate DP's automatic plug-in latency compensation by selecting it under Setup→Configure Audio System→Configure Studio Settings.

In the Soundbites window's Info pane, confirm that Use PureDSP Pitch Shift is the Transpose setting for the sound bites you plan to transpose (see Fig. 1). This is the default setting and will produce the most-natural-sounding results.

Now you're ready to tweak your guitar melody. First, in DP's Sequence Editor select the sound bites whose pitches you want to modify. Then choose the pitch mode for the sound bites by selecting Audio→Audio Pitch Correction→Set Pitch Mode For Selected Bites→Instrument. (Had your signal source been a singer, you would have chosen Vocals for the pitch mode.) Don't select Audio→Audio Pitch Correction→Set Track Pitch Mode, because this command will not affect sound bites that have already been recorded.

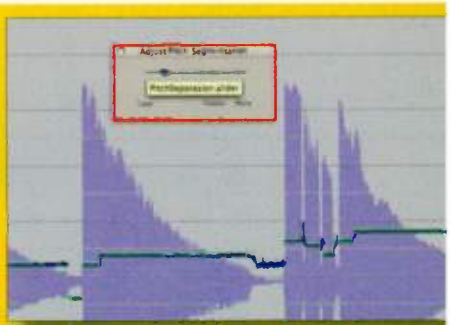
Next, choose Pitch in the track's track-layer menu. This action displays the blue

STEP-BY-STEP INSTRUCTIONS



Step 1: Select the sound bite you want to edit, then choose the proper pitch mode from the Audio Pitch Correction submenu.

Step 2: Choose Pitch in the track's track-layer menu.



Step 3: Adjust the degree of pitch segmentation using either the Pitch Separation slider or DP's Mute and Scissor tools.

pitch curve and green pitch segments superimposed over the waveforms of the sound bites. If you can't see the pitch curve, zoom in until it appears. You can move the pitch curve to a position above or below the waveforms by dragging the scroll handle in the track's Pitch ruler, which is located just to the left of the track's graphic display.

If the number of pitch segments is different from the number of notes played, increase or decrease the number of segments until there is one segment per note. A simple way to do this is to select the track, choose Audio→Audio Pitch Correction→Adjust Pitch Segmentation, and, in the window that appears, move the Pitch Separation slider to the right or left to increase or decrease the number of pitch segments.

Alternatively, you can click between two adjacent segments (spanning one musical note) with the Mute tool to join them together into one segment. Clicking on a pitch segment with the Scissor tool will split it into two segments.

Now for the creative part. Click-and-drag a pitch segment up or down to transpose the associated note's pitch in the same direction in half-step increments (see Web Clips 1 and 2). The note will maintain its intonation relative to concert pitch, remaining sharp or flat. To shift the pitch of any note in increments less than a half step, Command-drag its associated pitch segment up or down as needed. To move any note to its pitch center (so that it is neither sharp nor flat), select its pitch segment and choose Audio→Audio Pitch Correction→Quantize Pitch. If you want to revert an edited note to its original pitch, simply select its pitch segment and press the Delete key.

ONLINE BONUS MATERIAL

Sheer Genius or Garbage?

Because DP's pitch automation is applied nondestructively and in real time, you can bypass the automation on the affected track at any time during playback in order to compare your new melody with the original (see Web Clip 3). Toggle the P button at the bottom of the track's Pitch ruler to alternately bypass and activate pitch automation. To temporarily defeat pitch automation for one sound bite, open the Soundbites window's Info pane and choose Don't Pitch Shift for that sound bite's Transpose setting.

Once you're happy with your new melody, you may want to make it permanent by selecting any pitch-edited sound bites and choosing Audio→Merge Soundbites. Alternatively, bounce the edited parts (with pitch automation enabled) in real time to a new track. EM

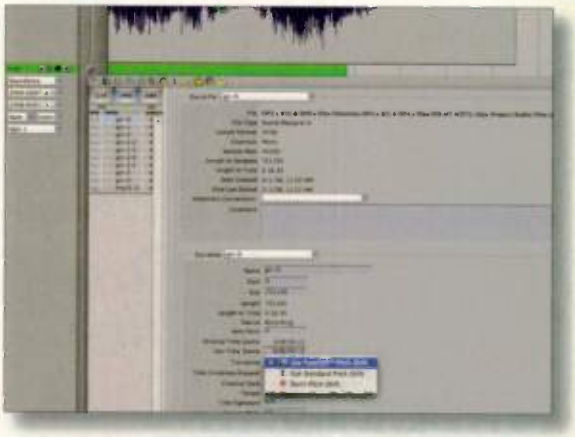
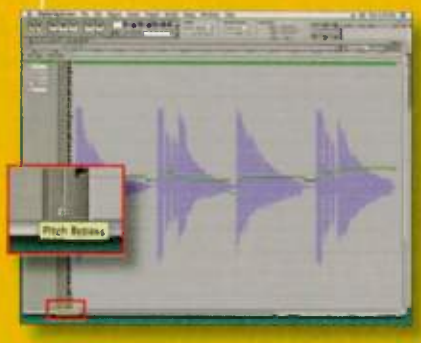


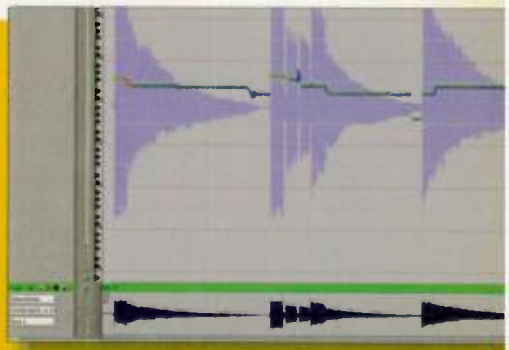
FIG. 1: Choosing Use PureDSP Pitch Shift for a sound bite's Transpose setting will yield the most-natural-sounding results when automating pitch transpositions.

EM contributing editor Michael Cooper is the owner of Michael Cooper Recording in Sisters, Oregon. Visit him at myspace.com/michaelcooperrecording.

Step 4: Click-and-drag pitch segments up or down to transpose their associated notes in half-step increments. Command-drag to fine-tune, or use the Quantize Pitch command to center the pitch exactly.

Step 5: Toggle the Pitch Bypass button on and off during playback to compare your edits with the original melody.





Step 6: To render your changes, merge the edited sound bite or bounce it to another track (shown here), with pitch automation enabled.

FIG. 1: The Euphonix MC Mix uses a protocol called EuCon to communicate with host applications via Ethernet more quickly and precisely than MIDI would allow.

Into the Ether

How Ethernet and OSC improve on MIDI. | By Brian Smithers

Musicians and engineers are always connecting one gadget to another, and as a result tend to develop a refined understanding of wiring standards, connectors, and, in the digital age, communication protocols. One such protocol, Ethernet, is showing up in an increasing number of audio devices. That fat telephone-style plug once used only for hooking up to a local area network is finding new and interesting uses.

The Lay of the LAN

Ethernet is the most common protocol for interconnecting devices on a network. It's a slight but useful oversimplification to say that the term refers to the RJ-45 (technically 8P8C) connector, Cat-5 cable, and packetized data format with which we're all familiar. Developed in the mid-1970s at Xerox, Ethernet offers speed, simplicity, and reliability. It also offers adaptability, as it is capable of carrying any sort of data, MIDI included. Although USB 2.0 and FireWire offer similar speed, availability, and flexibility, Ethernet allows longer cable runs.

Ethernet comes in three major types, defined by their bandwidth. 10Base-T offers 10 megabits per second of throughput, whereas 100Base-T and 1000Base-T offer 100 Mbps and 1,000 Mbps, respectively. 1000Base-T is also known as gigabit Ethernet. Ten-gigabit Ethernet is also defined, but it has not yet become commonplace. For comparison, FireWire 400 (IEEE-1394a) has a bandwidth of 400 Mbps, and USB 2.0 has a bandwidth of 480 Mbps.

It wasn't that long ago that conventional wisdom led the savvy DAW builder to remove any network devices from an audio computer. Since that time, however, network functionality has become ever more deeply integrated into the OS, and Ethernet ports have become ubiquitous on motherboards, including those in notebooks. Although one might still choose to keep a DAW disconnected from the Internet, local area networking has become an essential part of file and session management. Special servers can even allow streaming of audio and video assets across a network in real time.

Seizing Control

With a fast and proven data pipeline built into virtually every PC and Mac, it was inevitable that developers would start writing audio applications and creating devices to take advantage of it. Since the introduction of its first major control surface, the Pro Control, Digidesign has used Ethernet to exchange control gestures with Pro Tools. Even 10Base-T Ethernet, the stated minimum requirement for all Digidesign control surfaces, provides several times the 31.25-kilobaud bandwidth of MIDI, which is common in less expensive controllers.

This increased bandwidth allows the Pro Control, C|24, D-Control, and other Digidesign work surfaces to provide real-time metering, expansion to include dozens of faders, and reporting of track names and other information to the work surface. More speed allows more simultaneous control of level, pan, and plug-in parameters along with more-responsive writing of automation than is typical with MIDI controllers.

Similarly, Euphonix takes advantage of Ethernet's speed in its control surfaces, including the MC Pro and System 5 MC. A special protocol called EuCon translates the control surface's gestures into messages that can be understood by various host DAWs and that provide greater resolution than MIDI. This same arrangement has also been implemented in the more affordable MC Mix (see **Fig. 1**) and MC Control.

Either-net

Ethernet is also being used extensively to move MIDI and digital audio signals between audio devices. Distributed audio-playback systems in convention centers, theme parks, and other large installations often use one of various networking protocols—Cirrus Logic's CobraNet or Digigram's EtherSound, to name just two—to deliver dozens of streams of high-quality

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digital audio from centralized servers to background music systems. Performance venues are using these same protocols to replace multicore analog cabling in sound-reinforcement systems.

Roland's REAC recording system lets audio signals be digitized onstage and sent via standard Ethernet to a PC running Cakewalk Sonar to be recorded. An audio interface is required on the PC only for confidence monitoring.

Various products allow MIDI control of soft synths and synchronization of DAWs over Ethernet. Apple's Mac OS X has had this capability built in since 10.4 (Tiger) was released (see **Fig. 2**). MusicLab's MIDIoverLAN CP extends this concept to include both Macs and PCs in any combination.

Digital audio connections such as S/PDIF and AES3 let audio be shared between DAWs, but they require some sort of audio interface. Plasq Wormhole2 offers an open-source cross-platform way to send a stream of audio from one machine to another and back through a 100Base-T or gigabit Ethernet connection for processing on the remote machine. FX-Max offers FX Teleport, a Windows-only application that off-loads the processing of a VST plug-in running on the host DAW to a remote computer.

Muse Research has developed UniWire, a cross-platform method of communicating both audio and MIDI between the host DAW and the company's Receptor hardware plug-in host via standard Ethernet connections. This not only simplifies wiring, but also allows the plug-in to be instantiated from within the host DAW. Because both audio and MIDI are sent, both audio effects and virtual instruments can be controlled. Waves makes two hardware processors, the APA-32 and APA-44, that stream audio through Ethernet from a host computer to be processed by Waves plug-ins running on the processors' DSP chips. The plug-ins are instantiated within the host DAW's mixer, but they actually run within the hardware unit.

FIG. 2: This screen shot shows the configuration of a network MIDI port on a Mac. Since OS X 10.4, support for MIDI over LAN has been standard.

In sending MIDI and digital audio over Ethernet, manufacturers must be careful about introducing latency. Most such products not only are optimized for low-latency performance, but also provide ways of measuring and compensating for latency.

In Performance

Many of the most interesting current performance controllers are using Ethernet to provide fast and flexible communication of real-time gestures. Rather than transmitting MIDI messages, however, the Ethernet-enabled devices listed here use or can use a newer data protocol called Open Sound Control (we'll show you how to get two devices to communicate via OSC in next month's "Square One"). Developed and maintained by the UC Berkeley Center for New Music and Audio Technologies (CNMAT), OSC lets computers, instruments, and other devices share performance data across a network. OSC is optimized for real-time use and offers advantages in both speed and flexibility over MIDI. It is also extensible enough to allow for incorporation into applications ranging from James McCarthy's SuperCollider to MIT's Csound to Cycling '74's Max/MSP.

FIG. 3: JazzMutant's Lemur and Dexter controllers offer sophisticated, multitouch-screen gestural control over DAWs, soft synths, and performance applications.

> OSC offers advantages in both speed and flexibility.

JazzMutant's Dexter and Lemur controllers use a 100Base-T Ethernet connection to the host computer (see **Fig. 3**). This gives them ample bandwidth for the devices' multitouch display to convey multiple streams of real-time information. Both devices are supremely configurable, and OSC allows great latitude in determining precisely how to convey performance gestures.

IRCAM, the Institut de Recherche et Coordination Acoustique/Musique, has developed EtherSense, a device that digitizes the output of real-time performance sensors, such as motion, heat, and proximity sensors, and converts it to OSC. The OSC stream is then conveyed to one or more computers running compatible applications such as Max/MSP, Miller Puckette's Pure Data, or Native Instruments Reaktor. A single EtherSense unit supports up to 32 analog sensors, converts each of their readings into a 16-bit value at 500 Hz, and sends those values via 10/100Base-T Ethernet to a Mac or PC.

With Ethernet ports on virtually every modern computer, the opportunities for imaginative hijacking of what most people know as a networking protocol are almost limitless. Most of the implementations I've listed require only 10 or 100 Mbps connections—imagine the sort of applications that will take full advantage of gigabit Ethernet. One thing is certain: the days of removing or disabling a network card when building a DAW are gone for good. em

Brian Smithers is department chair of workstations at Full Sail University in Winter Park, Florida, and the author of Mixing in Pro Tools: Skill Pack *(Cengage Learning, 2006).*

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Q&A: Anne Cecere

Advice from a BMI expert on music for film and TV

"Film and TV is the new radio" is a mantra often chanted in industry circles today. Getting your music onto a hot TV show or composing for a film is the epitome of promotion, and it can be quite lucrative as well. Anne Cecere, BMI's associate director of film/TV relations (see Fig. 1), knows full well the important role that film and television can play in launching new artists and composers. She is charged with bringing writers into the BMI fold, working with music supervisors, and educating the industry and its newcomers on how to obtain these coveted placements. Cecere reveals here what every musician needs to know to pursue the "new radio."

By Fran Vincent

COURTESY ANNE CECERE

FIG. 1: Anne Cecere, BMI's associate director of film/TV relations, says that you can find music supervisors to submit your music to by searching the Web.

What's the first thing songwriters should do to get their music into film and television?

Do a little research and be strategic when it comes to sending your music out. Don't do a mass mailing to music supervisors. You can research from your home. Sit down and watch television. Start with the networks—ABC, NBC, and CBS—since they tend to pay the most when it comes to royalties. Cable pays great, too, but when you're starting out, try looking at the networks first. See what new shows NBC has. Then go to your cable stations. If your music sounds like it would be perfect for *Nip/Tuck*, then find the music supervisor for that show and send your stuff to him or her.

How do you find and contact music supervisors?

With Google and Internet Movie Database (imdb.com), it's very easy to find out whom to get your CD to. Every music supervisor has a different preference: some accept CDs, and some do not take unsolicited music. Others prefer MP3s. But most of them I deal with are surfing the Net and using MySpace and looking at certain music (sites) to see what's out there. Send a CD with a few specific songs targeted to the show, plus a letter of introduction referencing the TV series or film, and ask if they could give it a listen—no long backstories. And don't expect a call back. *Music Connection* magazine has a listing of all of the music supervisors that are in town (L.A.), and it will say if they take CDs. Another helpful tool is the *Film & Television Music Guide* by the Music Business Registry (musicregistry.com). It's like the Bible. We use it here at BMI. If you want to get into putting your music into film and TV or scoring, that is a great guide to have.

> Be strategic when it comes to sending your music out.

Should you call music supervisors?

No. A lot of times it's better to mail in your CDs. Research the music supervisor. A lot of them give interviews or have been on industry panels. Often you

can get the Podcasts of panels, so you can literally hear it from their mouths as to what their preference is.

What makes an artist more attractive to a music supervisor?

Your situation should be what a music supervisor calls a "one-stop situation." This means it takes only one phone call to license a song. So if I'm a music supervisor and I'm working on an episodic TV show and I literally have a day or two to license music for a scene, and I pull from my library three bands I think will all equally fit in the scene, the one that's a one-stop situation is most likely the one that will get my phone call. When you have no time, that one phone call to license 100 percent of the song can make all the difference in the world. When a music supervisor starts seeing multiple publishers they have to deal with, that can deter him from using that song. One-stop licensing is key, especially for episodic TV. Put ["one-stop licensing"] on your CD and jewel case and in your introduction letter.

Can a one-stop situation be formed when multiple writers are involved?

Sure. If there is more than one publisher on your song—let's say you are a songwriting duo or you're in a band and all of you decide to split the song's [rights] up evenly—you can create a one-stop by giving one person in the band the right to license the song [and, if possible, the recordings] on behalf of the other members. As long as you have an administration contract between the bandmates [or cowriters], you can consider yourself a one-stop.

A music supervisor is interested in using my song. What should I expect next?

When it comes to [licensing music for film and television], there are two licenses involved. The first is a sync license, which gives permission for your composition to be synced to a visual [a film or a TV show]. If you're an indie artist without a publisher, then the sync license will go to you from the film or TV producer. The other part of the equation is a master license, which is for use of the actual master recording. If you're an indie band and you're not with a record label or publisher, then you are the owner of the whole entire thing—composition and recording—so music supervisors are going to have to go through you for everything, making you a one-stop situation.

What if I don't have a lawyer to review these contracts?

There are organizations like California Lawyers for the Arts [calawyersforthearts.org] and Volunteer Lawyers for the Arts in New York [vlany.org] that provide legal consultations to artists at a reduced cost. Either one would be a great help. *The Musician's Business and Legal Guide* [Prentice Hall, 2007] also has samples of sync and master licenses.

What can an artist expect to make from a music placement for television and film?

There's no black-and-white. There are certain productions that don't give any money up front. Then there are others who have a really nice music budget, and they have money to pay you up front. Some productions have "most-favored nations" clauses in their contracts—meaning that whether you're Dylan or a garage band they just found online, everyone gets the same amount of money. In that case, there is no negotiation. For those that will negotiate with you, don't be a diva. If you've never had a film or TV placement, being on a hit show—whether they pay you up front or not—is probably going to be good for you for many reasons. Plus, even if you get no up-front money, you are still going to be paid performance royalties. For shows or movies aired on television in the U.S., you get paid per airing on each TV station. For theatrical showings of films, you are paid royalties for showings outside the U.S. only. [In the U.S., theaters are not required to pay performance royalties.] The royalty rates for both vary.

> There are certain productions that don't give any money up front.

How does an artist collect performance royalties on film and TV music?

First, you must join a performing-rights organization [in the United States, BMI, ASCAP, or SESAC]. A cue sheet should be given to the artist's PRO by the film-production company so we know what music was used in a film or TV show. That's how we know who to pay. We only accept cue sheets from production companies for legal reasons. If we don't have a cue sheet on file, the writer doesn't get paid. It's the writer's responsibility to make sure the cue sheet is on file with their PRO. Call us a few weeks after an airing to check that we have it. It takes about nine months from the airing to get paid your royalties. So don't wait until nine months pass before checking in. When you make a deal with a production company, ask them about the cue sheets. Say, "I know the PROs require a cue sheet in order for me to get paid. Who is in charge of that?" Then be sure to get contact information for that person. You, the writer, can ask for a copy of the cue sheet, but they aren't required to give you one.

What can a composer expect when asked to score a film?

It's a totally separate situation from licensed music placements. Ninety-nine percent of the time, the studio will own the publishing on the score. It will be a work-for-hire situation. To a novice, this seems unacceptable. I have to remind them that even the biggest film composers don't always get to keep their publishing. One exception is a production company who can't afford to pay an up-front fee. They may allow a composer to keep all or part of the publishing in lieu of money up front. But remember, you still get your performance royalties every time it airs on television or [is shown] internationally in theaters.

Any last words of wisdom for artists new to film and TV?

As a composer, don't underestimate student and indie films. And remember, no cue sheet means no performance money.

Fran Vincent is the author of MySpace for Musicians *(Thomson Course Technology, 2007) and president of Retro Island Productions, Inc., a music-marketing and consulting firm. Visit her at myspace.com/retroisland.*

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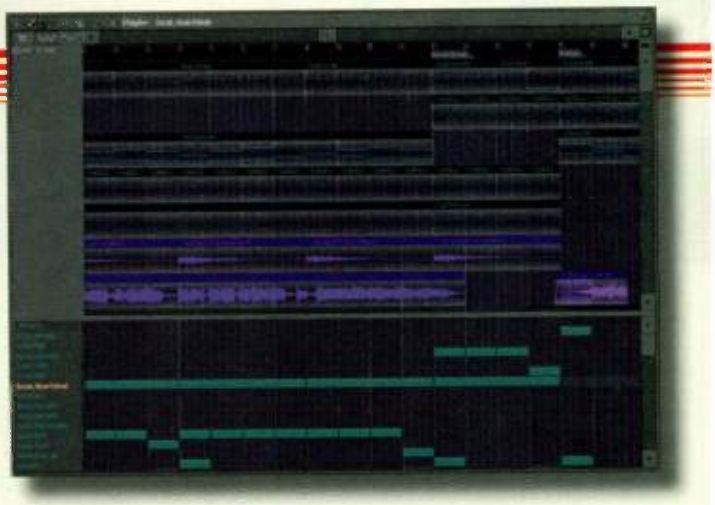

FIG. 1: FL Studio's Playlist view allows patterns, Audio Clips, and automation to appear in the same view. Note that the traditional block style for patterns is still available.

Image Line

FL Studio XXL 8 (Win)

This feature-laden package gains more functionality.

By Allan Metts

PRODUCT SUMMARY

digital audio sequencer $299 (download) $399 (boxed)

PROS: Lots of included sound generators and effects. Pattern-based approach fosters experimentation and creativity. Flexible audio and controller routing. Capable beat slicing.

CONS: Application can be confusing and non-intuitive at times. Windows can obscure one another in interface.

FEATURES	1	2	3	4	5
EASE OF USE	1	2	3	4	5
DOCUMENTATION	1	2	3	4	5
VALUE	1	2	3	4	5

Image Line
image-line.com

>> In our reviews, prices are MAP or street unless otherwise noted.

ONLINE BONUS MATERIAL

Image Line's FL Studio (formerly known as Fruityloops) has seen a constant evolution from a shareware pattern-based sequencer to a full-fledged environment for audio production. The program's ten-year evolution continues, with version 8 adding even more goodies.

FL Studio 8 is available in four different configurations, each with different feature sets and prices (three of these are available in either boxed or download form). Unlike with many competing programs, you're entitled to free updates for life when you purchase a license. I downloaded and installed the XXL version, which is the top of the line. After a quick online registration process, the software was activated and ready to go.

We've covered FL Studio several times before (most recently in the April 2006 issue, available at emusician.com), so I'll just review the basics and discuss the newest features. FL Studio combines pattern-based sequencing with audio recording, mixing, and a ton of great sound generators and effects. The term *pattern-based sequencing* probably doesn't do the program justice, however, because it's relatively easy to overlay linear tracks on top of the assembled patterns. The program supports ReWire, VSTi, and DXi instruments and DX and VST effects, and it can even provide multiple outputs to other host programs using ReWire, DXi, or VSTi.

Integration Station

One complaint about earlier versions of FL Studio was that the audio-recording features seemed tacked on to the original pattern-based environment, and I'm happy to report that Image Line is making strides to improve this. The Playlist view (see Fig. 1) now allows Clips containing audio, pattern, and automation events to appear side by side, with a common set of tools to manipulate all three (including slip edit capabilities, which are new to this version).

The familiar block-style representation of patterns, which lets you build a matrix of patterns as rows and bars with beats as columns, is still available and appears below the Clips pane. While I appreciate the flexibility, I found

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FIG. 2: The Edison audio editor is packaged as an effects plug-in, which lets you record and process audio anywhere you can place an effect.

the two independent representations to be confusing in practice—adding a Pattern Clip in the Clips pane doesn't add the same pattern at the same point in time in the Blocks pane. Only Pattern Clips have the ability to be resized and slip edited, so unless you're working with a file from an earlier version, I recommend simply ignoring the Blocks pane. There's an option to easily convert the old format to Pattern Clips if needed.

You get multiple ways to record audio as well. The original method (creating Clips directly in the Playlist view) is still there, but now FL Studio offers the Edison audio editor plug-in (see Fig. 2). Edison is a rather ingenious concept—think of it as a stereo audio editor embedded in an effects plug-in. The ingenious part is that this "effect" can actually record audio just like a standalone program. Simply insert Edison into an effects slot in FL Studio's mixer and connect that mixer channel to your audio source.

There are four modes of recording in Edison. It has its own set of transport controls, so the first mode is simply to press the record button and go. Two modes are input triggered and start recording when audio is detected (one of these stops the transport when the audio falls below a threshold level; the other doesn't). You can also instruct Edison to start and stop in sync with the main FL Studio transport.

Once you've recorded your audio, you can make any desired edits and send the edited work to the Playlist as an Audio Clip. And you don't necessarily have to record audio in order to use Edison. The plug-in also supports loading samples from disk.

As an audio editor, Edison covers the basics pretty well. You can do the slicing and dicing you'd expect, and there are provisions for fades, normalization, and loop tuning. It also offers convolution reverb, noise reduction, and a pitch-to-MIDI function that works pretty well on monophonic passages. You can see your audio in the spectral domain, but you can't directly manipulate the individual frequencies by mousing around in the spectrum. You can perform EQ operations on a time selection, however, which is a roughly comparable operation.

Slice the Beat

FL Studio already had a basic beat slicer, but the new Slicex generator puts the original one to shame (see Fig. 3). Slicex holds different waveforms in its two Decks and provides the requisite tools to divide the waves into regions either manually or automatically according to time, pitch, or transients. Once the regions are in place, you can sync them to tempo and assign them to individual MIDI notes. All of the waveform editing from Edison exists here as well.

That's just the start of the fun, however, because Slicex also includes tools to mix and mangle the regions in every way imaginable. Each region (in both Decks) can be given its own mixer channel, and each can be assigned to articulators controlling amplitude, a filter, playback speed, and sample start. There are eight articulators in all,

> Slicex should be just the ticket for creating constantly evolving dance grooves.

FIG. 3: The Slicex plug-in provides powerful beat-slicing capabilities. You can cycle between two different waveforms and modulate each slice with different parameters.

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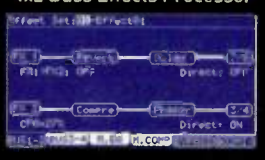

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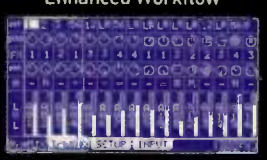
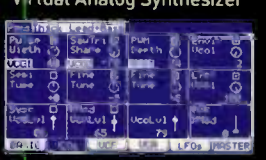


Virtual Analog Synthesizer



MPC5000
MUSIC PRODUCTION CENTER

Studio photo provided courtesy of Quad Recording Studios, New York City

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so you're likely to reuse them across regions.

Each articulator provides settings for its filter (filter type, slope, and so on) and a set of modulators. These modulators can control panning, level, cutoff frequency, resonance,

> The pattern-based approach can be a great way to foster the creative process.

playback speed, and the sample start position, and the source of this modulation can come from ADSR envelope generators, LFOs, the axes of an *x-y* controller, keyboard Velocity, or a random-number generator.

What's more, there are layering settings that control which Deck you're hearing. You can cycle between the Decks randomly or predictably, layer them both, or map them to one of the axes of the *x-y* controller. I loaded two complementary beats in the Decks, mapped the layer control and cutoff frequency to the *x-y* controller, and recorded to Edison as I tweaked the controls. You can hear the results in **Web Clip 1**.

Slicex is a lot of fun and should be just the ticket for creating constantly evolving dance grooves. For details on additional plug-ins and features new to this version of FL Studio, see the **online bonus material** at emusician.com.

The Big Picture

FL Studio certainly has plenty of functionality and flexibility, but there is a cost: at times I found the program to be confusing and non-intuitive. For example, its concept of channels is completely independent of what would

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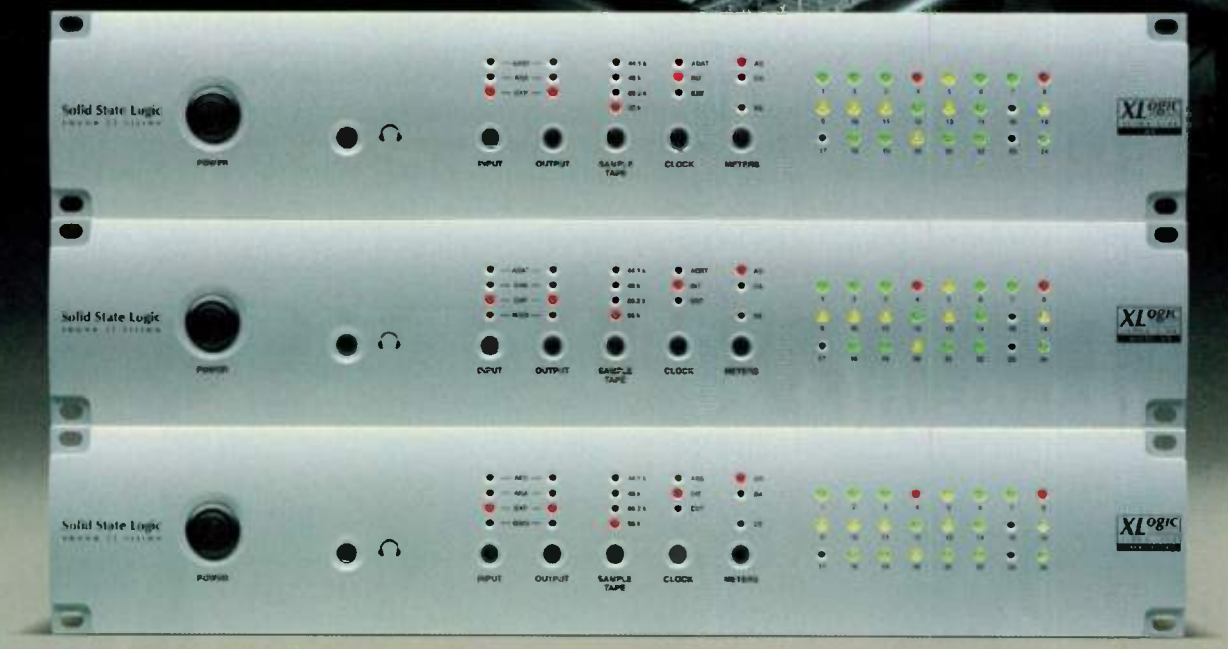

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- **Alpha-Link MADI AX (centre):** MADI ⟷ ADAT ⟷ Analogue
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> FL Studio has a reputation for extreme flexibility by not enforcing strict rules.

typically be considered channels in the Mixer view (here they are called Inserts). By default, adding a new channel in your FL Studio project does nothing in the Mixer—you must explicitly assign a Mixer Insert; otherwise, the audio simply gets summed in the Master channel.

When recording audio, the online help cautions against recording in the Master channel, yet this is precisely what will happen if you simply press the main record button and choose the recommended option of recording audio into Edison (and this recorded audio isn't visible in the Playlist until you put it there). I certainly prefer a one-to-one-to-one relationship between channels of audio source material, channel strips in the mixer, and tracks in the playlist. The concept of recorded audio existing only in the Master channel makes no sense to me.

FL Studio has a reputation for extreme flexibility by not enforcing strict rules about what connects to what. While experienced FL Studio users will probably feel right at home, first-time users may find themselves frustrated or confused by a program that doesn't follow the same conventions as other DAWs and that doesn't always constrain the user to choices that will achieve a desirable outcome.

I've reviewed many sequencers and digital audio programs, and I found myself going to the online help more frequently with this application than any other. Fortunately, the program is well documented, and video tutorials are available online (although as of this writing, most of the new version 8 features have no video tutorials).

Nevertheless, all of FL Studio's functionality represents a compelling value, especially when you consider Image Line's policy of free updates for life. I had a blast playing with the dozens of different sound generators spanning a wide spectrum of synthesis techniques. And the program's pattern-based approach can be a great way to foster the creative process, especially if you're into dance or techno music. Download the FL Studio demo and try it for yourself.

Allan Metts is an Atlanta-based musician, software/systems designer, and consultant. Check him out at ametts.com.

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FIG. 1: The Virus TI Snow's compact configuration of knobs and buttons provides quick access to programs and their essential parameters.

Access Music

Virus TI Snow

Virus TI technology comes to your laptop.

PRODUCT SUMMARY

hardware synthesizer	$1,550 (MSRP)

PROS: Extreme portability. Great sounds. Virus Control plug-in. ROM banks can be overwritten. Doubles as audio interface.

CONS: A few bugs in the software. A little pricey.

FEATURES	1	2	3	4	5
EASE OF USE	1	2	3	4	5
QUALITY OF SOUNDS	1	2	3	4	5
VALUE	1	2	3	4	5

Access Music
access-music.de

ONLINE BONUS MATERIAL

By Jon Margulies

Ever since the release of the first Virus in 1997, German synth manufacturer Access Music has been an industry leader in virtual analog synthesis. Not content to reproduce the sounds of yesteryear, Access has been busy raising expectations for modern-day synths. The newest Access instruments, the TI (Total Integration) series, have moved well beyond their roots, with modern features such as a grain-table oscillator and the Atomizer real-time audio processor.

The TI series pioneers a new level of software-hardware integration with Virus Control technology, which lets you use the synth as a plug-in within your sequencer while retaining the features of an independent piece of hardware. The latest addition to this series is the Virus TI Snow, a portable unit with the complete sound engine of its older siblings.

The Snow may well become a favorite synth of laptop performers and mobile producers (see **Web Clips 1 through 4**). Its lower price is going to grab the attention of anyone who hasn't previously been able to afford a Virus, and it even comes with a carrying case. So does the diminutive Snow live up to its potential?

Working the Hardware

This synth is a small 11 × 2 × 6–inch desktop unit with a solid feel (see Fig. 1). It has six knobs and 21 buttons, sports the same wintry white-and-gray look as the 37-key Virus TI Polar, and has the same black-and-white LCD. Of the six knobs, one is the master volume and two are hardwired to filter parameters. You configure the remaining three (called Soft Knobs) to control any parameter.

The Snow is surprisingly easy to edit given its small number of physical controls. Presets are organized in eight banks, which you access with two button presses. Once a bank is selected, you can call up any of its 64 patches by typing in the patch number using the two rows of numbered buttons. Anyone using the

unit in a live setting will appreciate being able to directly jump to patches this way.

Once a patch is selected, you use the Soft Knobs for basic sound tweaking. In the factory presets, you'll find them doing anything from adding effects to mangling multiple complex parameters. The two knobs to the right of the three Soft Knobs always control the frequency and resonance of Filter 1. When you hold the Shift button, these five knobs control part parameters: volume, transposition, panning, filter envelope, and amplitude envelope.

I found it aggravating that twisting the knobs brought up a window displaying the parameter name and value. The window obscures all other information on the screen, including the labels of the Soft Knobs. A quick trip to the Config menu lets you turn this feature off, but I think the ideal solution would be to display a smaller window that leaves more of the screen visible.

The Snow doubles as an audio and MIDI interface featuring 2-in/2-out, 24-bit audio.

FIG. 2: The Virus Control plug-in is an essential tool for anyone wanting to do serious sound design on the Snow.

> The Virus Control plug-in is the best way to get inside this machine.

Using the Virus instead of an additional sound card makes your portable setup significantly more powerful without adding much weight. Unfortunately, that setup is not suitable for DJ-style cueing because the Snow lacks a second pair of stereo outs.

Under the Hood

You can access any of the Snow's 12 edit menus with two button presses, and the parameter editing setup makes the most of the three Soft Knobs. By default, the edit menus present you with the three parameters deemed most important for the patch you're editing, each mapped to one of the Soft Knobs. For example, if the patch uses the Classic oscillator type, you'll see Shape and Wave Select in the oscillator edit menu, whereas if you're using a wavetable oscillator, the same knobs will control Index and Interpolation.

If you prefer to do more-detailed editing, all parameters are available at the touch of a button, or you can make expert editing the default mode. However, for serious synth programmers, the Virus Control plug-in is the best way to get inside this machine (see Fig. 2).

If you're already familiar with the Virus synthesis architecture, you'll feel right at home programming sounds on the Snow (see the **online bonus material** at emusician.com). The only difference between it and the other TI models is that it uses a single DSP chip instead of two, which translates into lower polyphony (a maximum of 50 voices) and 4-part multitimbral capability instead of the other models' 90 voices and 16 parts.

Fortunately, the sound engine uses processing power dynamically, reducing polyphony as the complexity of the patch increases. Therefore, it's important to identify the more resource-intensive aspects of the sound engine. The *Quick Start* guide is full of useful information on optimizing your patches, and I advise new users to take some time with it.

The Software

The Snow runs the latest Virus TI software, version 2.7.0, which was still in beta as of this writing. Its new features are the Atomizer real-time loop processor, an additional stereo USB output (bringing the total to three), support for multiple Viruses connected to the same computer, and the Virus Control Center application.

Virus Control Center is a standalone utility you use to back up and restore the contents of the Snow's memory (see Fig. 3). You also use it to burn patches into the ROM banks, which, until now, could not be overwritten. At first I was unable to get Virus Control Center to communicate with the unit, but a restart of both the computer and the Snow did the trick.

FIG. 3: Virus Control Center lets you burn sound banks into the Snow's ROM. It can also perform backups and update firmware.

The Virus Control plug-in is a pleasure to use. It has a well-designed interface that makes programming clear cut and much simpler than some soft synth interfaces I've seen. You can really forget that you're working with a hardware synth and operate as if it were a soft synth, but with a huge savings of CPU power.

Atomizer

Atomizer is a welcome and unexpected new feature. It works in parallel with the synthesis engine, and its focus is processing audio rather than producing it. When Atomizer is enabled, the Snow passes incoming audio directly through to the outputs, using its Beat Scanner to determine the tempo of the incoming material. (You can also tap in the tempo.)

Incoming MIDI notes determine Atomizer's action. Notes E1 to B1 (MIDI Note Numbers 40 to 47) sample and loop the incoming audio at different rates, D1 reverses the sample, and C#1 and D#1 gate and filter the incoming audio as controlled by the MIDI Modulation Wheel. Notes C2 and above produce high-speed loops tuned to the note's pitch. In an interesting twist, the Snow's effects processor is engaged only when a loop is triggered. So you could be playing a flanged and bit-reduced 16th-note loop, then with a single keystroke return to the unaffected audio. When Atomizer is active, you can use the bank select buttons to trigger the loops and gates, so you can use Atomizer without an external keyboard.

If you're an experienced synthesist, you're probably already familiar with the sounds the Snow can make. From classic and huge to weird and futuristic, there's something here to satisfy most sonic tastes. Even with its convenient front-panel controls, the Snow is not going to be the most appealing synth for those who want row after row of knobs. But for users who make the majority of their music on their laptops, the Snow is an incredibly attractive package. I see it joining my UAD-Xpander in making my two-year-old MacBook a production powerhouse.

Jon Margulies is a producer, guitarist, and DJ in New York who has been performing professionally since he was 11 years old. You can catch up with Jon and his many projects at heatercore.net.

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FIG. 1: Mu Voice's interface features an area for editing Chord Schemes, a 4-channel mixer, and an area for managing presets.

Mu Technologies

Mu Voice 1.2 (Mac/Win)

A great-sounding plug-in for instant harmony.

By David E. Weiss

PRODUCT SUMMARY

harmony-generating software	$279

PROS: Easy setup. Excellent sound quality. Powerful interface for automating presets and chords. Easy to manipulate with a MIDI keyboard. XML-based settings for compatibility.

CONS: Limited effects. Contrapuntal arrangements difficult to orchestrate.

	1	2	3	4	5
FEATURES			3		
EASE OF USE				4	
AUDIO QUALITY				4	
VALUE				4	

Mu Technologies
mu-technologies.com

ONLINE BONUS MATERIAL

Those hoping for new developments in the vocal-processing market have reason to rejoice, for a new voice has entered the choir. Mu Voice 1.2 is a plug-in that tunes and harmonizes your vocal tracks. It is the flagship product of Belgium-based Mu Technologies and is delivered in VST and RTAS formats for the Mac and PC and AU format for the Mac.

Mu Voice uses a proprietary algorithm recently developed at Antwerp University in Belgium. This algorithm is distinctive in that it analyzes audio in extremely small units, which reduces latency and gives higher resolution in the time domain. Because there is a lot of variation in pitch in every note that even the most accomplished singer produces, large analysis windows create artifacts, called smearing, when the same frequency is assumed over too long a time. This is analogous to the way that a high-resolution digital photograph shows fewer artifacts than a low-resolution photograph because the low-resolution photograph uses fewer pixels, which smears the same color over a larger area.

Lift Up Your Voice

Mu Voice is easy to use, and I found it fun, inspiring, and instructive—fun because it's immediate, inspiring because the harmony voices are very realistic, and instructive because when you slide off-key, so do your virtual backup singers, greatly enhancing the tiniest error. Enable it on a vocal track, pick up a mic, and choose a preset (I recommend starting with a trio), and you'll hear several clones of yourself singing along in harmony in different ranges (see **Web Clips 1 and 2**). You can try this out in the free, time-limited version available from the Mu Technologies Web site.

As with any vocal-processing software, Mu Voice performs best when it's processing a clean signal, so put it first in the chain if possible, and certainly in front of any reverb or delay. Occasionally the harmonizing voices were garbled beyond recognition, but clicking on the Inversion button in the Analysis settings quickly fixed that.

Mu Voice has a 4-channel mixer with which to control the source voice and three harmony voices (see Fig. 1). By default, the

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FIG. 2: The MIDI keyboard panel replaces the area below the faders and lets you issue commands or enter notes and chords.

second voice sings a third above the first, the third voice sings a fifth, and the last voice sings an octave. But it's easy to change the pitch of any voice within up to two octaves above or below the default pitch. With a bit more tweaking, you can change the pitch to anything you desire (more on that later).

Each channel strip has Volume, Pan, and Mute controls, as well as a Humanize slider to introduce subtle variations in pitch and vibrato and a Formants slider to adjust the timbre and character of a voice. Two drop-down menus (Filter and H.EQ) offer a variety of effects. Mu Voice's Pan slider delivers a bit more depth than left-right balancing; the manual says that the slider's parameters are based on measurements relative to the listener's nose, and to my ears it has a full, three-dimensional quality.

The Formants slider sets the contour of frequencies that characterize the timbre of a voice. You can tweak this curve without affecting a note's pitch. Moving the Formants slider to the left expands the formant spectrum, producing a more nasal, squeezed-down effect (similar to the *e* in *ear*). Moving the slider to the right narrows the formant spectrum, producing a more open sound (similar to the *o* in *olive*). The middle position preserves the formant spectrum of the lead voice.

I found Mu Voice's controls very useful for smoothing out harmony parts. As noted in the review of Antares Audio Technologies Harmony Engine and Zplane.Development Vielklang (see the April 2008 issue of EM, available at emusician.com), synthetic harmonies can sound a little unnatural when soloed. Though harmonization software is not intended for this, soloing the parts helps you make subtle improvements (see Web Clips 3 and 4).

Strike a Chord

Beyond creating harmony voices at regular intervals, you can make Mu Voice follow specific chords: major, minor, diminished, sixths, sevenths, ninths, and suspended fourths. You use the two Chord menus to define chords for the source melody. One cycles through the available key signatures, and the other lets you choose a chord from your chosen signature. I found that awkward and a bit mind-numbing. Fortunately, Mu Voice also lets you select the key from your MIDI keyboard. Better yet, play a chord and Mu Voice will capture it for you.

Once you have the chords, you play the piece through in Write mode, indicating chord changes with your mouse or your MIDI keyboard. Then, in Read mode, Mu Voice will dutifully follow the song, moving from chord to chord at the appointed place.

> You can make Mu Voice's harmonies follow specific chord changes.

Calling the Shots

The chord scheme is where Mu Voice gets interesting: a cell does more in Mu Voice than define the key and chord for a certain passage; it is a snapshot of all of Mu Voice's settings at that point. For example, if you've designated that a certain passage be linked with a C major chord, you can also tell Mu Voice to apply a certain preset. (A preset defines a wide variety of characteristics, from the presence or absence of an EQ filter to how many voices are singing and in which octaves.) Mu Voice comes with 36 presets, which you can overwrite without erasing the defaults, and it lets you save an additional 24 of your own. Projects, which contain both Chord Schemes and settings, are saved as XML files, and they are compatible with any version of Mu Voice on any platform.

In a useful touch, a preset can shift the pitch by as much as 12 semitones up or down. That lets you use Mu Voice to create counterpoint; however, it requires a lot of mental gymnastics. Mu Voice doesn't visually represent the separate voices in a piano-roll-style editor as does Vielklang or Celemony Melodyne. Both Vielklang and Melodyne require a separate process of analysis prior to creating harmonies, whereas Mu Voice operates in real time.

Because Mu Voice does its work very quickly, analyzing the source melody and producing harmonies with a latency of as little as 5.8 ms, it is well suited for live performances. Onstage, a MIDI keyboard opens up highly expressive possibilities. By pressing a key, you can call up any preset, jump to any spot in the Chord Scheme, and make your virtual backup singers change key (see Fig. 2).

Playing for Effect

Mu Voice comes with two effects offering a limited number of settings. (The company says more control will be added in future versions.) Filter lets you dial up nine different filter types: highpass and lowpass, which attenuate the frequencies above and below 4,000 Hz with a 6 dB slope; a telephone-line effect, which filters frequencies between 400 and 4,000 Hz (see Web Clips 5 through 7); and comb and inverse-comb at multiples of 300, 600, and 1,200 Hz. The combs are most interesting—imagine waves rolling through an EQ, affecting groups of frequencies.

Harmonic EQ is based on Mu Voice's core algorithm. It lets you control the amplitude of the individual harmonics of a given sound rather than changing the amplitude of a fixed frequency range. For instance, it might alter the tenth harmonic of a given note, no matter what that note is. Harmonic EQ settings follow the melody. (Compare Web Clips 8 and 1. You'll notice that in the Harmonic EQ version [Web Clip 8], it sounds as if the voice were singing from inside a bottle.)

Though its deeper features could use refining, Mu Voice offers immediately gratifying, great-sounding harmonies, and it performs exceptionally well in real time. It is a welcome new addition to the pantheon of harmonization software, and it can only get better as it evolves.

David E. Weiss is a songwriter, musician, and self-producer in the San Francisco Bay Area.

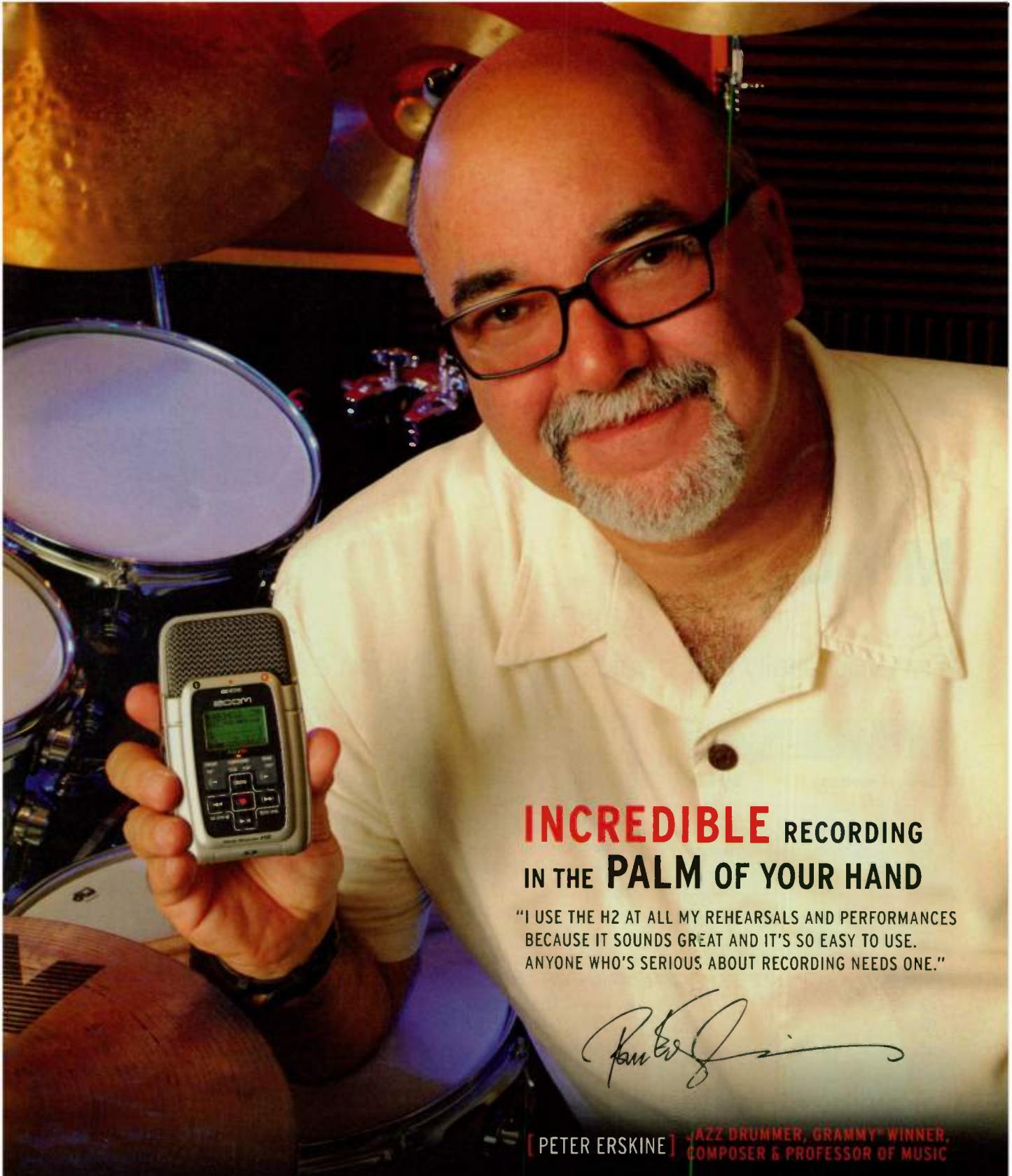

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FIG. 1: Fresh industrial design, an 8-inch woofer, and a new tweeter waveguide make the BX8a Deluxe the best Studiophile monitor to date.

M-Audio

Studiophile BX8a Deluxe

Can you really get champagne sound on a beer budget?

By Jason Blum

PRODUCT SUMMARY

powered monitors	$499/pair

PROS: Improved styling, high-end response, and stereo imaging. Plenty of power. Outstanding value.

CONS: No auto power switching. No EQ options. Boomy, undefined bass.

	1	2	3	4	5
FEATURES				●	
AUDIO QUALITY			●		
VALUE					●

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ONLINE BONUS MATERIAL

In recent years, M-Audio has focused a great deal of effort on becoming arguably the best-selling manufacturer of studio monitor speakers. The fruits of that labor have paid off handsomely in the latest BX-series monitor, the Studiophile BX8a Deluxe—a new and improved version of the BX8a, which debuted three years ago and replaced 2002's BX8. With a redesigned high-frequency waveguide, an 8-inch Kevlar woofer, 130W of biamplified power, and frequency response rated from 40 Hz to 22 kHz (±2 dB), the specs are almost too good to be true in such an affordable pair of monitors.

Recently I put a pair of the new Studiophiles through their paces in my studio. Though I found some performance trade-offs at such a low price point, M-Audio has indeed made some noteworthy improvements over prior generations.

Out of the Box

Sold only in pairs, the BX8a Deluxes are presumably close-field monitors, but their large woofers and considerable 26-pound heft make them an imposing sight when I place them closer than 6 feet from my listening position. The MDF (medium-density fiberboard) construction feels solid and is free of any rattles or buzzing, and the matte vinyl lamination gives the monitors a warm, burnished, professional look and makes them a bit less slippery during handling. A single bright-blue LED indicates when the monitors are getting juice. Aside from the BX8a and M-Audio logos across each monitor's front panel, I found the sleek and minimal aesthetics to be a big improvement over the angled-cabinet design of the first-generation BX8 (see Fig. 1).

The rear panel is relatively spartan compared with the original BX8's, too, sporting XLR and balanced ¼-inch inputs along with a sole volume knob for trimming gain on the internal amps, a standard IEC C14 power connector, and a rocker-type power switch (see Fig. 2). A large bass-reflex port vents to the rear a few inches below the monitor's top edge. Conspicuously absent from the monitor's back side are any sort of EQ-adjustment tools, which I found a bit strange, as the old first-run BX8 monitors I've used in my studio DJ rig since

FIG. 2: The BX8a Deluxe lacks any EQ trim controls, offering only a volume knob and power switch on its sparse back panel.

2002 feature boost and cut switches for high, mid, and low frequencies. The lack of such tuning facilities on the BX8a Deluxe isn't a significant issue, but it does strike me as a step backward in flexibility.

Power Plant

Two onboard amps drive the BX8a Deluxe's dual drivers, with 60W dedicated to the 1.25-inch silk-dome tweeter and another 70W flowing to the woofer (both ratings are continuous power into 4Ω). With ample power available to each speaker, a pair of BX8a Deluxes can kick out some serious volume. Each driver is shielded from transient audio spikes by a protective circuit that keeps it safe from harm during loud sessions or the occasional hot-patched audio cable.

Whenever I audition a new pair of monitors, I always turn to tried-and-true music I know like the back of my hand. My reference stable includes a cross-section of tracks from multiple genres, from classic rock and ambient to pop and progressive house, all auditioned through a Lavry DA10 D/A converter with Zu audio cables. Every pair of monitors reacts differently to this battery of tests, and a few tracks always stand out as ideal tests for each particular set. In this case, five tracks—Quivver's "Surin," Pink Floyd's "Comfortably Numb" and "Mother," Future Sound of London's "My Kingdom," and the fourth movement from Beethoven's Ninth Symphony—offered ample evidence of the BX8a Deluxe's strengths and weaknesses.

The monitors acquitted themselves admirably on nearly all fronts in all genres, displaying a remarkable aptitude for wide stereo imaging and detail. The silk-dome tweeter, improved crossover, redesigned waveguide, and dedicated 60W amp all added up to a remarkable level of high-frequency detail that's quite unusual for speakers selling for this price. I was particularly impressed with the way the monitors rendered the wide, out-of-phase stereo ambience that opens FSOL's "My Kingdom." Lush reverbs swirled around without washing out details in high-end percussion, and every instrument felt firmly rooted in the stereo field. Likewise, the mid and high frequencies in Quivver's progressive house tune "Surin" were crisp and sharp—almost bordering on shrill, but not fatiguing in any way. The upper registers of the kick drum were firm, solid, and punchy.

The two Floyd tracks further illustrated the BX8a Deluxes' aptitude for great imaging. Roger Waters's voice on "Mother" was firmly planted center stage, and when the full band kicked in, the wide stereo spread on guitars and piano produced an enveloping sensation that rivaled that of monitors clocking in at twice the price. The vocal chorale in Beethoven's Ninth felt a little less focused, and although the live orchestral nature of the recording didn't translate as well as the studio recordings, the speaker still produced a full and satisfying listening experience.

Loose Caboose

The BX8a Deluxes offer an admirable amount of high-frequency detail and great stereo imaging, but they didn't perform quite as well on low-bass frequencies, exhibiting a tendency to translate them as loose and boomy. The woofers delivered tons of punch in the upper bass registers, but sub-bass frequencies simply didn't have that same tangible sense of presence and accuracy. I listened to "Surin" and found the kick punchy and tight with plenty of chest feel, but as soon as the bass line kicked in, the region between 60 and 120 Hz muddied up and gave the overall mix a boxy sound. I noticed the same phenomenon while listening to "Comfortably Numb"; highs and mids were bright and focused, but during chorus segments, when the bass guitar hits hard around 100 Hz, those low frequencies boomed out of the monitor without clear definition and obfuscated details that are so important in that area.

I also noticed that the monitors sounded far better at moderate levels than when pushed to the limit. That's not to say they can't rock the house—they're quite capable of earsplitting volume—but at some point, they lose detail across the board and become a mushy wall of sound. Coupling the BX8a Deluxes with a sub would no doubt work wonders here, giving the internal amps a little extra breathing room at higher volumes.

Now Hear This

M-Audio has always had a handle on getting great products to market at budget prices, and the BX8a Deluxe continues that tradition by delivering great imaging and high-frequency detail without denting your life savings. That bargain price does include some trade-offs—bass performance is, understandably, not up to par with that of some pricier monitors, and the speakers don't offer any onboard EQ options—but with a street price of just under $500 per pair, they're an unbeatable set of starter monitors for getting a project studio off the ground.

Established studios with high-end close-fields will likewise find these monitors a worthy addition as a louder close- to midfield supplemental set, particularly when coupled with a matching subwoofer. Few monitors deliver this level of performance at such a reasonable price, so if you're in search of a pair of powerful budget monitors, be sure to put the BX8a Deluxe at the top of your audition list.

Best known as a DJ and producer in the electronic group Deepsky, Jason Blum is currently focused on commercial mixing and mastering in his Los Angeles studio.

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SAMSON

GM1U G-Track

By Rusty Cutchin

The G-Track combines a large-diaphragm condenser microphone with a digital audio interface that offers connections for instrument and line inputs and monitoring by headphones.

At first glance, the Samson G-Track large-diaphragm condenser mic and USB audio interface ($149.99) looks like one of those gimmicky hybrid devices that do many things but few of them well. But looks can be deceiving; thanks to an ingenious design, the G-Track could be a perfect fit for the traveling singer-songwriter who wants to record ideas on the road.

MIC MATTERS

The upper part of the G-Track looks like a standard side-address large-diaphragm condenser. The mic employs a 0.75-inch, 3-micron diaphragm and has a supercardioid polar pattern. It includes neither a rolloff switch nor a pad. A/D conversion is 16-bit, at 48 kHz.

The capsule is covered by a heavy wire-mesh grille that stands up well to daily use. On the rim, where the head meets the mic housing, are two slide switches. The Input switch toggles between mono instrument as well as microphone and stereo line inputs. The Direct Monitor switch selects among three headphone-monitor modes: Mono, for monitoring the mixed mic and instrument inputs and computer playback; Stereo, for accurate monitoring of a stereo line source mixed with computer playback; and CPU, for monitoring computer playback only.

Below the switches is a column of three knobs that control playback level, instrument-signal level, and mic level. These spring-loaded knobs can be pushed down and locked flush with the housing to keep your settings from being accidentally changed.

> I was pleasantly surprised by the G-Track's sound quality.

The mic's main output is the USB connector; there is no XLR output. The USB port is flanked by two stereo ⅛-inch jacks: a line/instrument input and a no-latency headphone output that can also feed a mixer or powered monitors. The jacks are placed so that you can have a permanent setup, with all cables bundled neatly together as they descend from the mic.

The mic can be mounted on the included desktop stand with swivel adapter or on a standard floor stand, and an optional shockmount (the SP04; $39.99 [MSRP]) is available. In addition, Samson provides cables for almost every conceivable setup, including a headphone extender, guitar cable, 10-foot USB cable, and stereo mini to dual RCA splitter cable with RCA to ¼-inch adapters. These extras add greatly to the G-Track's convenience.

INTERFACING THE MUSIC

I wasn't expecting a lot from a combined mic and interface at this price, so I was pleasantly surprised by the G-Track's sound quality. The mic worked well on my neutral baritone voice, which was enhanced by the bump at 3 kHz; singers with brighter voices might not fare as well. Furthermore, the G-Track proved to be a good choice for narration as well as for vocals.

The G-Track requires Windows XP, Windows Vista, or Mac OS X 10.4 or later and comes with a copy of Cakewalk Sonar LE. Samson does not bundle audio software for the Mac, presumably because so many Mac users already have GarageBand. I tested the unit using OS X 10.5.2 (Leopard), and my Power Mac G5 immediately recognized the G-Track interface.

The interface uses basic USB audio codecs on Macs and PCs, providing two tracks of recording and stereo playback. Using MOTU Digital Performer 5.12, I was able to reserve my FireWire interface for output while assigning the G-Track to two input channels, testing it with vocals, guitar, and stereo synthesizer. You can use two G-Tracks for stereo recording by selecting two USB devices in Windows. Certain Mac DAW applications may require creating an aggregate device in OS X.

The G-Track had ample headroom on inputs and outputs, driving powered speakers as well as headphones. It delivered plenty of gain in Mic, Instrument, and Line modes.

The mic's frequency response has a bump in the range from 3 kHz to 10 kHz and a less pronounced bump up to 20 kHz, so it isn't a great recording mic

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PRESONUS

Monitor Station

By Allan Metts

The PreSonus Monitor Station expands your studio's monitoring muscle with four headphone amps, a talkback section, and the ability to switch between three input sources and three speaker pairs.

PreSonus has a reputation for producing great-sounding audio products at reasonable prices, and the Monitor Station ($399.95) doesn't disappoint. Housed in a rugged chassis that sits on a table or desktop, the 8 × 10–inch device provides headphone amps, speaker switching, input source switching, and talkback capabilities.

The Monitor Station lets you connect as many as three stereo sources—two on balanced ¼-inch connectors and one on RCA jacks—and route them to three pairs of speakers. The RCA input has its own level control, as well as a phono preamp and a grounding screw for connecting a turntable. The Monitor Station uses a wall wart for power.

You choose input sources and speaker pairs using large, illuminated buttons, which can be configured in either toggle mode (making one choice cancels the others) or combination mode (all choices can be active simultaneously). If you use a subwoofer, you can leave it connected while you toggle between two sets of mains. Toggle mode is great for quickly switching between speaker pairs while auditioning your mix. In addition, you can either sum the inputs or toggle them one at a time. Using toggle mode with input sources is handy for comparing your mix to commercial CDs.

CALIBRATION STATION

Each of the three speaker selections has an adjustment knob for calibrating its level. The manual provides extensive instructions on calibrating your system. For accurate comparisons, most users will want to calibrate each set of speakers to the same loudness level.

The Monitor Station provides no amplification and requires powered speakers or speakers with an outboard amplifier. In addition to my main studio monitors, I typically check mixes on my computer's external speakers and on consumer-grade speakers. Because all of the Monitor Station's speaker outputs are on balanced ¼-inch TRS connectors, I had to dig through my adapter drawer to get everything connected.

Once you've calibrated your speaker

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for singers with bright upper registers, but it will make serviceable demos for most performers, especially songwriters looking to get ideas down.

ON THE G-TRACK

The surprising heft of the G-Track, along with its ample included accessories, makes the unit feel like a professional product. Keep in mind that while the sound quality is quite good for a mic at this price, the G-Track is not intended to be a substitute for a professional recording mic. It's a good choice for the new computer-based recordist who is adding narration to video tracks or creating Podcasts, and it's a logical tool for remote voice recording with a laptop.

Considering its price and the convenience of having a combined mic and interface, the G-Track will be a good investment for a large number of singer-songwriters who need to record on the go.

Value (1 through 5): 4
Samson
samsontech.com

PSPAUDIOWARE

Xenon 1.0 (Mac/Win)

By Richard Alan Salz

PSPaudioware's Xenon mastering plug-in ($249) includes essentials such as a 2-stage limiter, an integrated leveling amp, absolute intersample peak distortion prevention, and word-length reduction (requantization) with three noise-shaping options. Also on hand are K-System metering, 64-bit processing, and support for files with up to 192 kHz sampling rates. The plug-in runs under AU (Mac only), VST, and RTAS hosts; I tested the VST version on a dual-Xeon workstation running Magix Sequoia 10 and Steinberg Nuendo 4.

EM HOT PICK

Xenon uses the ubiquitous iLok system for copy protection. Registering Xenon at PSPaudioware's site was straightforward, and the iLok authorization was ready to download in about 2 hours.

It's easy to get the hang of Xenon when you realize that its first stage of limiting provides most of the gain reduction that will take place. The input and transient controls determine the initial threshold of limiting, much like an analog limiter would. Xenon's second stage uses look-ahead-based brickwall envelope detectors to eliminate the possibility of digital overs. Next in line is a switchable option to enable the envelope detector to prevent intersample peak distortion in the digital-to-analog converters that will ultimately play back the material. This feature works by monitoring the digital sample values and preventing multiple full-scale samples from occurring.

Reminiscent of some popular legacy analog devices, Xenon includes a leveling amplifier that is really nice for adjusting the macrodynamics of the material being mastered. In effect, this is a final gain-reduction stage with a very slow time constant that prevents the limiter from "pumping" as it recovers from a loud section.

A-OK

Mastering engineer Bob Katz developed the K-System of metering, which is a multiformat display that shows true RMS and peak levels as well as crest factor simultaneously. Once you get the hang of it, the system helps you understand the concept of loudness in a more concrete way. The K-System also allows for three metering variants/scales based on commonly accepted monitoring levels, which are calibrated to broadcast (12 dB), CD production (14 dB), and film production (20 dB). Speaking of calibration, Xenon includes an onboard pink-noise generator that makes setting up levels relatively quick and easy.

Although Xenon is mainly intended for use in the mastering stage, I also found it very useful in other applications. It can do amazing things to an ambient drum track that would normally require some specialized hardware. For example, I'm a fan of the sound of radical limiting, especially when combined with trashy-sounding sources, and Xenon allowed me to quickly and effectively dial that effect in. The included leveling stage keeps the sound constant without the annoying pumping that's often the hallmark of extreme processing. Xenon let me produce a polished and dynamic sound faster and more easily than some other mastering-oriented plug-ins I've used.

I did find that it took a little while to get used to the K-System metering. Once I did, however, I was left feeling that conventional metering leaves quite a bit to be desired. Whether due to the 64-bit internal processing or just the overall software design, Xenon was free from digital-sounding artifacts, such as a perceived loss of depth or stereo image quality. Though it is possible to make sources sound flat and lifeless (yet still loud), even fairly aggressive processing can yield a natural-sounding result. My kind of plug-in!

Xenon delivers the goods when it comes to mastering-quality limiting at a very low cost. Superb sonics, a nicely designed interface, excellent value, and groundbreaking metering combine to make Xenon an easy product to recommend. Prospective purchasers who already have an iLok should take note that PSPaudioware offers a fully functional 14-day demo on its Web site.

Value (1 through 5): 5
PSPaudioware
pspaudioware.com

PSPaudioware's Xenon offers a range of mastering tools, including K-System metering, a 2-stage limiter, and word-length reduction. It supports 64-bit processing and files with up to 192 kHz sampling rates.

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levels, use the large level knob in the unit's center to control all outputs simultaneously. Just above the knob is a stereo LED peak/level meter with eight steps; you can configure it to peak at +4, +10, or +18 dBu. To either side of the level control are buttons for muting and summing stereo signals to mono. The Monitor Station retains its speaker, input, and LED meter configurations when you power it up.

RIGHT ON CUE

The speaker outputs, LED meters, mute and monitor buttons, and main level control all use the main output bus for audio. A second bus labeled Cue adds substantial flexibility during recording. The Cue bus has its own set of input selectors and a separate level knob, making it easy to provide a click track or an alternate mix that doesn't appear on the mains.

On the back panel are dedicated ¼-inch TRS outputs for both the main and cue signals. I found the Cue bus to be most useful in conjunction with the headphone amplifiers. The four headphone jacks are at the top, just above their respective level knobs and source selection buttons. Each headphone circuit can connect to either the main or Cue bus, allowing your drummer to hear the click track, for example, while the vocalist hears only the mains. My AKG headphones require a hot signal, and the Monitor Station delivers. The headphone amps sounded clean, loud, and clear. In fact, I could detect no coloration or induced noise anywhere in the Monitor Station's audio signal.

TALK BACK

The Monitor Station provides a talkback channel with a dedicated level control and a built-in microphone on the front panel. The button that activates the circuit can be latching or momentary, and it glows bright red when it's engaged. If you prefer, you can connect your own talkback microphone to an XLR connection on the back panel. The Monitor Station offers no phantom power, however.

The talkback signal appears only in the Cue bus, and the bus's signal diminishes automatically when you activate the channel—a nice touch. A dedicated Dim knob controls how much the signal drops, and you can activate the Dim function independently of talkback using a dedicated button.

All in all, the Monitor Station is a great little unit that performs admirably. If you could use some additional monitoring flexibility in your studio, the Monitor Station may provide just what you need.

Value (1 through 5): 4
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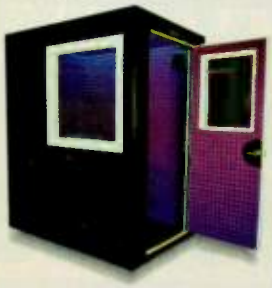
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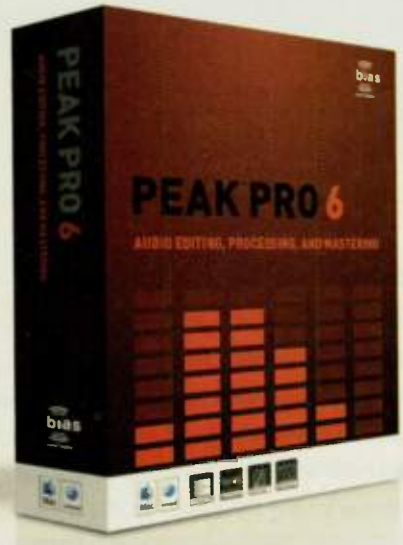

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Good Times, Bad Times

By Nathaniel Kunkel

Wow, things are just crazy these days. Banks are failing and people are losing their money, with lines around the block and everything. (Unnecessary lines, as the FDIC is properly funded, but lines nonetheless.)

Even more amazing is that engineers and producers are willing to admit they aren't working. That wasn't the case two years ago. You would go to an industry mixer and it seemed everyone was working. In reality, of course, we were not. It's been tough for a while, but we felt that looking busy was the best way to *be* busy. However, it seems that it's too grim to lie about it anymore. Way too grim to feel happy about how cool Melodyne is. (Actually, that's not true: I am so digging Melodyne. I just cannot believe what that program can do. And when Direct Note Access comes out, it's all over but the crying.)

Okay, so things could be worse, but they sure could be better. So just in case no one else has the cojones to say it: it's hard for everyone in the music business right now. If you're struggling, you're not alone. If you're working, you might want to keep that to yourself. Man, times *have* changed.

"Don't despair," my friends tell me, "even in the Great Depression, people wanted to be entertained."

"Yeah, but back then, they didn't have as much stored media at home," I counter. I also think the fact that people weren't buying music before the economy tanked isn't a good sign.

This is where I would usually present the silver lining. Not this time. I think it's going to suck for a little while longer.

Remember that the problem with the music business is the same as with the housing market: it's a broken model and it needs to be restructured. It's a drag when that comes after a failure. It's cooler when you catch it early.

Maybe the music industry's problems are a huge blessing. I mean, there has been some cool stuff released, but for the most part, while the proliferation of Pro Tools has put the power of production in everyone's hands, it doesn't make them instant songwriters.

Like I've said before, maybe if the music business weren't so lucrative, all the people who are only in it for the money would go away and leave it for those of us who would do it for free anyway. We could then make a higher concentration of good art and, hopefully, some money again.

Hey, there's the silver lining. Yeah, that's the ticket: we'll work for free but the majority of music that people are exposed to won't suck anymore. Wait—is that better? Ugh, maybe we should all start lying again.

You see, the record business has always been cyclical. Major record companies get huge, lose track of the art, and fail. Indies get bigger, make more profit than a major, then get bought by major record companies that ride for a bit on the coattails of the people who actually knew what they were doing, but then fail because they weren't responsible for the success in the first place. Sound familiar?

So don't despair. Although it's not getting better right now, it is going to get better. em

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Nathaniel Kunkel (studiowithoutwalls.com) is a Grammy and Emmy Award–winning producer, engineer, and mixer who has worked with Sting, James Taylor, B.B. King, Insane Clown Posse, Lyle Lovett, I-Nine, and comedian Robin Williams.

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